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SOFTSWITCH DESIGN AND PERFORMANCE ANALYSIS

Master's thesis for the degree of Master of Science in Technology submitted for inspection in Vaasa, 26th of April, 2010.

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ABBREVIATIONS AND ACRONYMS

2G	Second Generation Mobile Communication System
3G	Third Generation Mobile Communication System
ATM	Asynchronous Transfer Mode
AUC	Authentication Centre
API	Application Programming Interface
AS	Application server
AS-F	Application Server Function
A-F	Accounting Function
AGS-F	Access Gateway Signaling Function
ANM	Answer Message
ACM	Address Complete Message
APM	Application Transport Message
AMR	Adaptive Multi Rate
BSC	Base Station Controller
BSSAP	Base Station System Application Part
BISUP	Broadband ISDN User Part
BICC	Bearer Independent Call Control
BCTP	Bearer Control Tunneling Protocol
CNCS	Core Network Circuit Switching
CAMEL	Customized Application for Mobile Enhanced Logic
CA-F	Call Agent Function
CSF-N	Call Service Function Node
CC	Call Control
CS	Circuit Switched
CORBA	Common Object Request Broken Architecture
DTMF	Dual Tone Multi Frequency
EIR	Equipment Identity Register

FLA	Fully Layered Architecture
FNR	Flexible Numbering Register
FDM	Frequency Division Multiplexing
GSM	Global System for Mobile Communication
GMSC	Gateway Mobile Switching Centre
HLR	Home Location Register
ISDN	Integrated Service Digital Networks
IXCs	Interexchange Carriers
ISUP	Integrated Service Digital Networks User Part
IW-F	Interworking Function
INAP	Intelligent Network Application Part
IAM	Initial Address Message
I-BIWF	Initiating Bearer Interworking Function
ISC	International Softswitch Consortiums
IPCC	International Packet Communication Consortium
LECs	Local Exchange Carriers
MSS	Mobile Softswitch Solutions
MSC-S	Mobile Switching Centre Server
M-MGw	Mobile Media Gateway
MAP	Mobile Application Part
MGCP	Media Gateway Control Protocol
MG	Media Gateway
MIN	Mobile Intelligent Network
MGC-F	Media Gateway Control Function
MG-F	Media Gateway Function
MS-F	Media Server Function
MTP	Message Transfer Part
MSCP	Mobile Service Control Point
OSA	Open Service Architecture

RAN	Radio Access Network
R-F	Routing Function
R-BIWF	Receiving Bearer Interworking Function
OMAP	Operation Maintenance and Application Part
OoBTC	Out of Band Transcoder Control
O & M	operation and management
PSTN	Public Switched Telephone Network
PLMN	Public Land Mobile Network
PCM	Pulse Code Modulation
PAM	Pulse Amplitude Modulation
PBX	Private Branch Exchange
SS7	Signalling System 7
SPC	Stored Program Control
SG	Signaling Gateway
SIP	Session Initiation Protocol
SCE	Service Creation Environment
SLEE	Service Logic Execution Environment
SIGTRAN	Signaling Transport
SMS	Short Message Service
SMC	Service Message Center
SSF	Service Switching Function
SC-F	Service Control Function
SPS-F	SIP Proxy Server Function
SG-F	Signaling Gateway Function
SCCP	Signaling Connection Control Part
SDP	Session Description Protocol
SLA	Single Layer Architecture
SMSRLA	Subscriber Management and Signaling Routing Layered Architecture
SMR	Subscriber Management and Routing

TDM	Time Division Multiplexing
TG	Trunking Gateway
TCAP	Transaction Capabilities Application Part
TUP	Telephone User Part
TSC	Transit switching Center
VOIP	Voice over Internet Protocol

SYMBOLS

τ	Time Delay
λ	Mean arriving rate
μ	Mean service rate
\tilde{N}	Mean number of customer in the system
ρ	Offered traffic
T	Mean waiting time in the system

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ABSTRACT:

The increasing number of subscribers' demands in telecommunication sector has motivated the operators to provide high quality of service in cost effective way. Moreover, operators need to have an open structure system so that they can move their systems to the next generation network architecture. For this purpose, Softswitch is an appropriate technology because it is a safe and cost efficient solution and though it can migrate from traditional circuit-switching based telephone system to internet protocol packet-switching based networking. Softswitch network divides the logical switch into several parts with different functions such as signaling gateway, media gateway, media server, etc. Standard communication protocols are implemented between those parts. Softswitch is software-based system to make connection between devices, and moreover to control voice calls, data and routes calls through different entities of the networks. Softswitch supports management functions such as provisioning, fault handling and reporting, billing, operational support, etc. Softswitch suitable for all types of traffic and services so it is very demanding in the competitive world of mobile operators.

In this thesis, Softswitch has been studied and analyzed in details. Softswitch network consisted of different integrated modules such as transportation, calling and signaling, service application and management. Each module provides different services such as call control, routing, billing and network management. Each module is discussed from functional and service point of views. Softswitch based wireless network architecture as well as variety of service solutions is presented. Different protocol interfaces in softswitch network such as signaling system number 7 are explained. Moreover, bearer calls, independent call control protocol, gateway control protocol, IP bearer control protocol are explained as well. Variety of softswitch network architectures analysis has been done based on their performance and the applicability. Three Softswitch network architectures are proposed which are validated through simulations.

KEYWORDS: Softswitch, Public Switch Telephone Network, Mobile Softswitch Solution, IP Network, Signaling System 7

1. INTRODUCTION

The International Softswitch Consortiums (ISC) defines Softswitch in simple terms as follows: “A Softswitch is a software-based entity that provides call control functionality.” The term softswitch is a generic name that came with a new switching system idea. Softswitch network low costs means that compared with a local telephone system, the migration to a softswitch network needs end to end packet of voice services. This packet system involves digitizing, compressing and dividing voice in packets. Whereafter, the packet can be sending various routes from the sender to the receiver. Softswitch runs many types of devices, such as many types of gatekeepers, media gateways, call agents, access devices and Internet Protocol (IP) terminals that control signaling and protocols. A gatekeeper is part of a VOIP service and its task is to convert the voice and signaling from analog PSTN, SS7 to an IP packet. It is the task of the call agent to manage the call control function. The PSTN was designed primarily for voice communication but, increasingly service providers are looking for a single network that will able to provide voice, data and other multimedia facilities.

The core part of PSTN is Class 4 and Class 5 switches cannot be used for data and other multimedia supports. Because of the switches of their software and hardware are so tightly integrated that it requires not only software but also hardware changes. Softswitch is a software-based switching system that runs on standard hardware and replaces the central office switching functions. Softswitch can perform the same functions as the traditional switches and provides better, faster and cheaper solutions. Softswitch also provides facilities for translating different signaling protocols and permits PSTN calls to be relayed to any packet switch network. In other words, it provides the Next Generation Network (NGN) applications demands such as video conferencing, chat and browsing that can be delivered from IP servers.

1.1 Mobile Softswitch Solution (MSS)

Mobile Softswitch Solution (MSS) is mapped out according to Layered architecture network design which separates both physically and logically. Service management and controls (Control Layer) and transport service data layer (Connectivity Layer). MSS can work on a mobile network which applies Core Network Circuit Switching (CNCS). MSS works jointly through two different nodes one is MSC server (MSC-S), which is in the control layer, and the other in the Mobile Media Gateway (M-MGW) that is in the connectivity layer. Figure 1.1 shows the Mobile Softswitch Solution Architecture (Ericsson AB 2006:11).

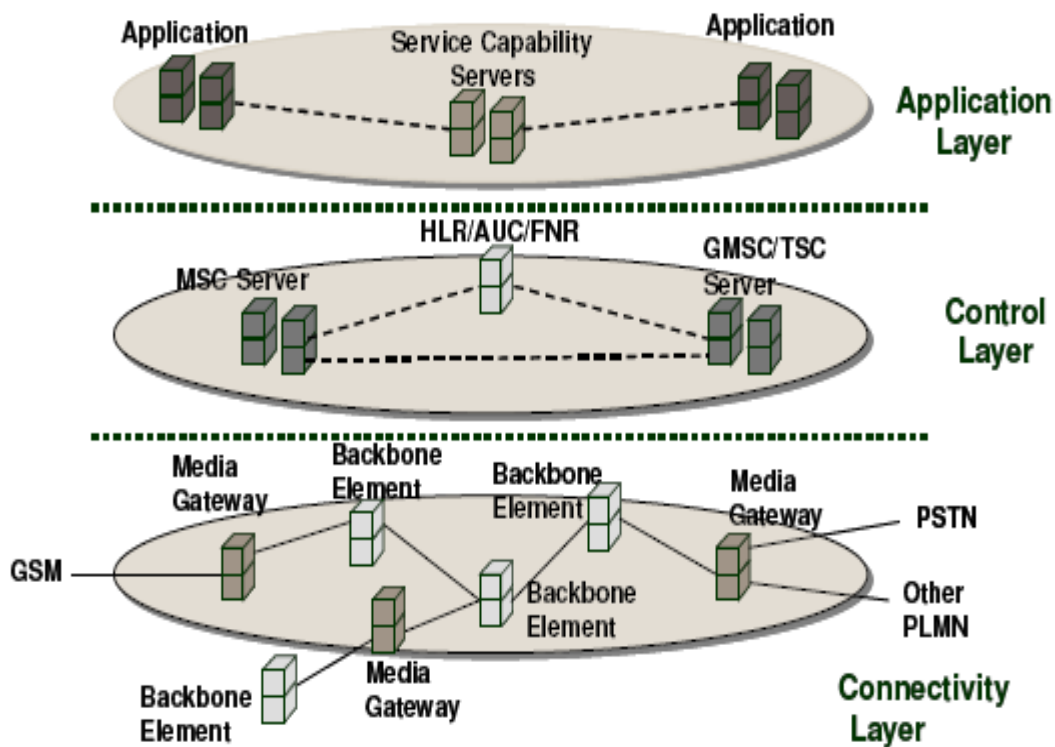


Figure 1.1 Mobile Softswitch Solution Architecture (GSM mobile softswitch solution R3.1 introduction 2006: 12).

1.1.1 Connectivity Layer

The MSS connectivity layer works through an internet protocol that helps to make end to end connections throughout the core network. The M-MGw tasks are to handle the call processing and transport traffic to the Circuit Switched (CS), which helps to establish the connection through the external network as PSTN or PLMN (Ericsson AB 2006:11-12; Alja & Brodnik 2004:159-160).

1.1.2 Control Layer

The control layer is the core part of the softswitch which performs call control and routing operations. In softswitch-based wireless networks, softswitch work as an integrated feature of Mobile Switching Center (MSC). The MSC server takes care of all types of network signaling and monitoring of CS calls. The control layer's task is to perform the control and analyze the traffic through the circuit switched networks, and to use standardized signaling to control the allocation of required resource in the connectivity layer (Ericsson AB 2006:11-12; Alja & Brodnik 2004:159-160).

1.1.3 Application Layer

The application layer is located on the top of the control layer. Different types of application servers are present in the application layer. The main task is to create a Service Creation Environment (SCE) and Service Logic Execution Environment (SLEE). SCE is a graphical user interface where the user can develop the system easily after that the SLEE compiles the program and establishes communication with the control layer through the Application Programming Interface (API) (Ericsson AB 2006:11-12; Guizani 2004:361-362; Xu, Su & chen 2003: 2744; Lemay, Suryn & Brown 2006:3299-3300).

1.2 Why Softswitch

The next generation wireless network should create a powerful service environment with superior flexibility, because of the rampant increase of the subscribers and their service demands. Therefore, it is important to construct a powerful service environment over the wireless network, which is not provided by the existing wireless network. Softswitch provides a flexibility advantage with simplified management network and services in the same infrastructure when compared to the PSTN. Since Class 4 and Class 5 switches were developed only for voice services, it is not possible to develop new services. Moreover, software and hardware are so compactly integrated that new services require not only the software, but also the hardware (Xu et al. 2003:2744; Ohrtman 2002:129).

In the year of 2000 boom was bust because new technology came into the market known as Competitive Local Exchange Carriers (CLECs). The failure of CLECs resulted in a net investment loss of trillions of dollars badly affecting capital markets as well as severally depressing the overall telecommunication economy. The more expensive part is the purchasing and maintaining cost of Class 5 and Class 4 switches. Whereas Class 5 and Class 4 switches are used for local and long distance provider accordingly. Those switches are really expensive to purchase and maintain and require very large and expensive space (Ohrtman 2002:2).

Six years after the passing of the telecommunications Act 1996, the fully setup of Class 4 and Class 5 could handle only nine percent of total telecommunications of the American residential phone lines. The primary problem was that the telephone company needed to access to the Class 4 and Class 5 where the lines were connected through the copper wire. After that wireless service became very popular throughout the world and required an alternative switching architecture. The monolithic telecommunication structure problem occurs when the central hubs of the PSTN is located to the natural disaster area or attacked

by the terrorist. In 11th September when the world trade center was attacked by the terrorist, five central office of the Verizon (the largest telephone company) was very near to the world trade center. Where more than 6 million data line passed to switching centers and almost 3.6 million data circuit and 10 cellular towers were destroyed. After long time new technology has come that provides low cost alternative than Class 4 and Class 5 switches. To reach the market demand telecommunication and data communication play combined action (Ohrtman 2002:2-4).

1.3 Benefits of Mobile Softswitch Solution

The key benefits of Softswitch are (Ericsson AB 2006:14; Ohrtman 2002:114-116; Ohrtman 2004:68).

- It can handle very high volumes of traffic, which is why there is a migration from circuit switching to packet switching.
- It has a layered architecture and that is why there are different functions in different layers as they are easy to maintain.
- Softswitch equipments take less space, reducing the footprint.
- Softswitch provides the latest technology servers and gateways, which ensure very high level of performance, and reduces the size and power consumption.
- For transmission, it uses less bandwidth because of packet transmission.
- It provides an open architecture, which can meet future demands.
- It provides many services to same transport network
- Softswitch allows the service provider to integrate their system with a third party application, and if necessary, they can write their own interpreting PSTN via SS7.

2. SWITCHING SYSTEM

In 1884 approximately 350,000 telephones were connected with the help of manual switching system. This was a simple concept whereby a switchboard operator manually connects one part of a pair to the other part to connect two phones. Strowger's Potential Company was the first to develop an automatic switching system around 1896. It was operated through pulse dialing, generated by the telephone device. In the year of 1958 Bell labs introduced the Stored Program Control (SPC) system. SPC provided different types of service such as call forwarding, call waiting and billing. It could also handle large volumes of calls in the peak hours, approximately 750,000 very efficiently, and provided a better service (Kularatna & Dias 2004:99-100; Ohrtman 2002:10).

The arrival of the semiconductor memory digital system made the design of the switching system very cost effective. This is one reason why developers used a digital switching system based on Pulse Code Modulation (PCM), where the voice divided into samples. In the years from 1980 to 1990s most telecommunication operators replaced their analog switching with digital switching systems. From 1980 to 2000, digital switching systems have been supporting about 800 million PSTN customers around the world. Moreover, they have been supporting mobile internet, multimedia, and broadband services (Kularatna & Dias 2004:99-100; Ohrtman 2002:10).

2.1 The public switch telephone network

The Public Switch Telephone Network (PSTN) constitutes three layers access, service and infrastructure. Telephone sets are connected to the access network with copper wire. When the user wants to connect to another user in the service node or switching node they dial the number, where after it goes to the nearest switching system that maintain some information

about the particular user interprets signaling information and makes a connection path. PSTN works as a star network.

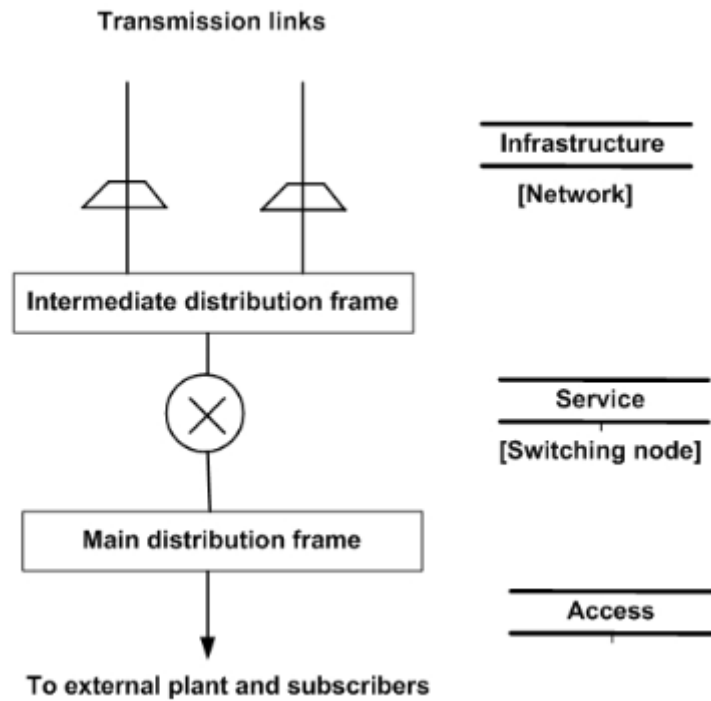


Figure 2.1 Three layers of PSTN (Essentials of Modern Telecommunications Systems 2004:100).

Nowadays the switches are part of a real-time processor-based system which operates the central element that controls the three subsystems of PSTN (Kularatna & Dias 2004:100-101; Ohrtman 2002:10-12).

2.2 Class 4 and Class 5 Switching

Class 4 and Class 5 switching play a core part of PSTN. Figure 2.2 shows the relationship between Class 4 and Class 5. When a subscriber tries to call long distance, first the phone

set connects through the Class 5 switch. A point to be noted that the Class 5 switch handle local calls, after that the calls goes to the Class 4 switch, which handles long distance calls. Depending on the situation, the call may be routed through other Class 4 switches. After that, the call terminated to the Class 5 switch and then the phone set. The design of Class 4 and Class 5 switches takes 25 years to come into service (Ohrtman 2002:13). The Nortel DMS-250 Class 4 switch is a very popular product as is the 4ESS from Lucent technology for local call handling, such as Class 5.

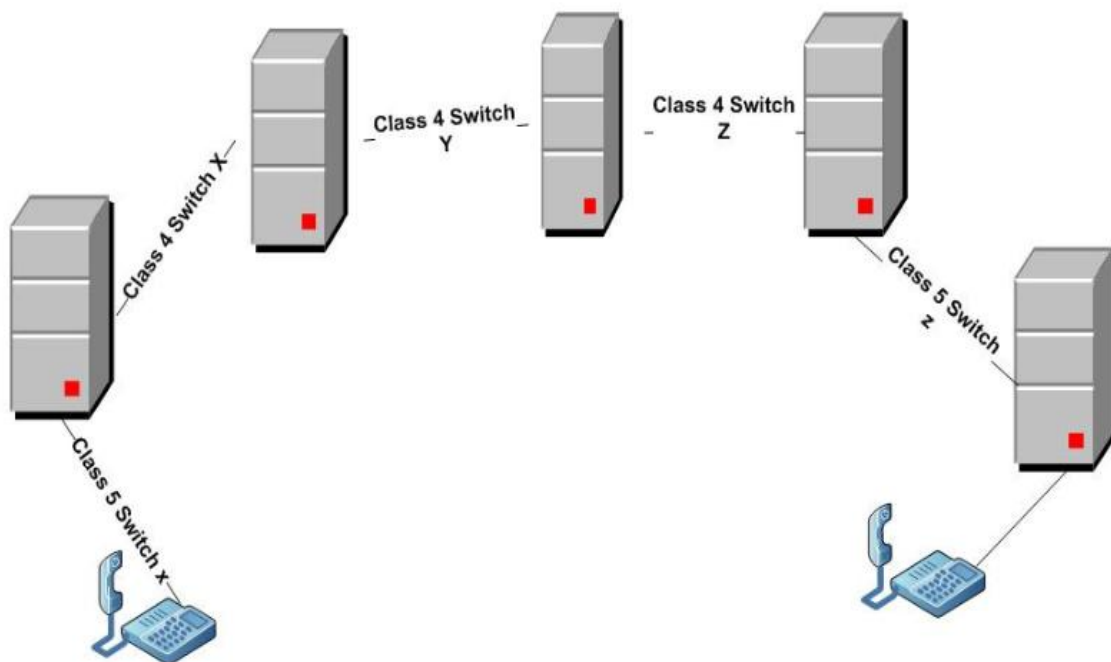


Figure 2.2 Relationship between class 4 and class 5 (Softswitch 2002:13).

“DMS-250 hardware, for example, is redundant for reliability and decreased downtime during upgrades. It has reliability rating 99.999 % (The five9s), which meets the industry metric for reliability” (Ohrtman 2002:14). The main hardware of the DMS-250 system includes the DMS core, switch matrix and trunk interface. The DMS core is the Central Processing Unit (CPU) and memory of the system managing high-level call processing, system control function, system maintenance and installation of new switch software. The Switch matrix switches the calls to their destination. The trunk interface peripheral modules

that establish a bridge between the DMS-250 switching matrix and trunks. DMS-250 real time manages customer-billing transactions and generates calls detail records automatically (Ohrtman 2002:12-14).

2.3 Private Branch Exchange (PBX)

Many companies use Private Branch Exchange (PBX). They connect their telephone lines to the PBX system and the PBX connected to the PSTN. PBX reduces the number of required lines and reduces the cost. Otherwise, large number of telephone lines would be needed to connect to the PSTN (Chava, Ilow 2007). PBXs systems are computer based and enable some soft changes to be made through an administration terminal or PC. (Ohrtman 2002:14-15).

2.4 Multiplexing

Multiplexing is the process where all incoming signal are combined and process into a single signal then transmitted. The method of making most effective use of available channel capacity is called Multiplexing.

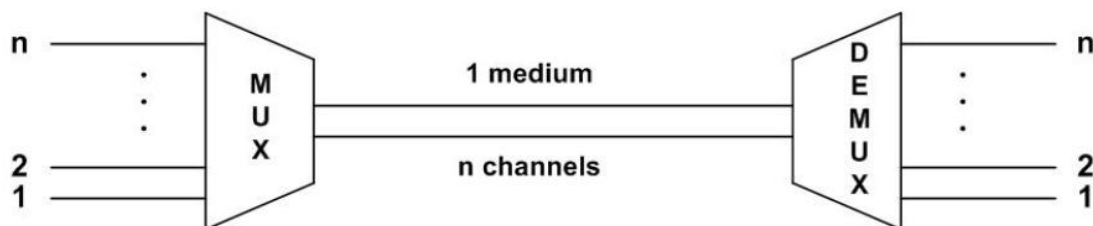


Figure 2.4 Concept of multiplexing (Scribd 2008).

Figure 2.4 shows simple procedure of multiplexing where data are combined through 1 to n in the multiplexer. Afterward, send it through data link and before send to the appropriate

output line it uses demultiplexer (Audestad 2007:73). Different types of multiplexing techniques are described next.

2.4.1 Frequency division Multiplexing (FDM)

In frequency division multiplexing the total system bandwidth is divided into the number of users. Figure 2.4.1 shows the FDM. For example, if the total system bandwidth is 60 KHz and each user share 20 KHz which means there are 3 users can share the same circuit. It is used in analog Radio and TV transmission. The oldest multiplexing technology is FDM. This was the only technology available before digitalization (Ohrtman 2002:16; Audestad 2007:74).

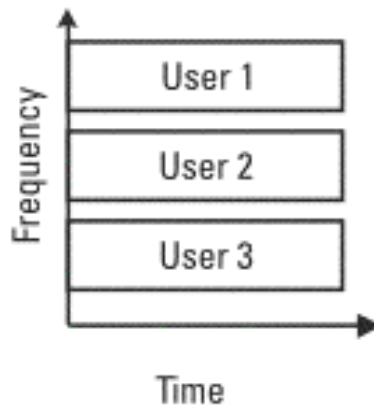


Figure 2.4.1 FDM (Interference Analysis and Reduction for Wireless Systems 2002:133).

2.4.2 Time division Multiplexing

The improvement over FDM was Time Division Multiplexing (TDM). TDM was applicable practically after semiconductor devices came to the market around 1950s and 1960s. Figure 2.4.2 shows TDM where using different time but same frequency. TDM is a digital transmission scheme. A repeater, known as a regenerator can receive weak and noisy digital signal. Remove the noise, reconstruct the original signal and amplify if before transmitting to the next segment of the transmission (Audestad 2007:74; Ohrtman 2002:16).

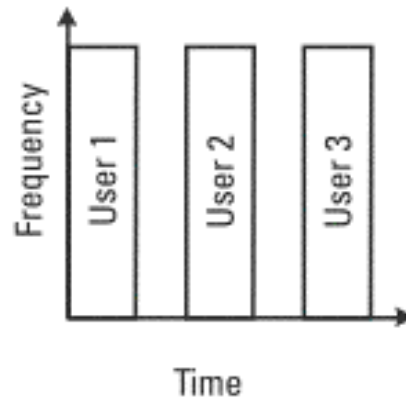


Figure 2.4.2 TDM (Interference Analysis and Reduction for Wireless Systems 2002:133).

2.5 Pulse Code Modulation (PCM)

“When the information source is analog (microphone, video signal, temperature sensor, etc) it is needed to convert it to digital format. The conversion process consists of two main parts, the first part is the discretizing of the signal and taking samples. Then the second part is the quantizing process of these samples to digital format” (Elmusrati 2009: Lec 4, 21). Pulse Code Modulation (PCM) is the standard digital format for voice signals. As an example, Figure 2.5 shows carrier multiplexer which uses 24 analog channels and each channel has 8000 sample/sec. The Nyquist criterion that the minimum sampling rate required to reconstruct the original signal from a set of uniformly spaced discrete time sample is $f_s \geq 2f_m$, where f_m is the maximum frequency component in analog signal and f_s is the sampling frequency rate. Sampling rate is very important, if using higher sampling rate then this means wasting of bandwidth. Otherwise, if the sampling is lower than needed aliasing will be happened and this will distort the original signal (Winch 1998:21-24; Ohrtman 2002:16-18; Shepard 2001:203-205; Sklar 2004:63; Elmusrati 2009: Lec 4, 26).

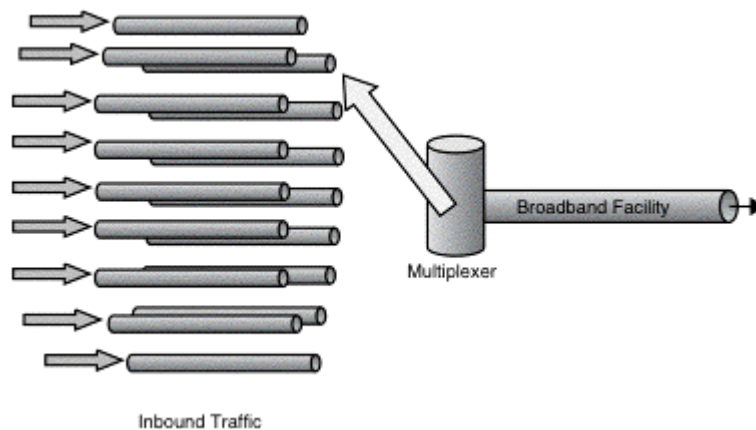


Figure 2.5 Time division Multiplexing (Telecom Crash Course 2001: 203).

There are four steps to complete the Pulse code modulation (PCM) they are as follows

- Pulse Amplitude Modulation (PAM)
- Companding
- Quantization
- Encoding

2.5.1 Pulse Amplitude Modulation (PAM)

This is the primary stage of PCM where analog signal represents digital bit streams. Figure 2.6 shows the process of Pulse Code Modulation Technique (PCM).

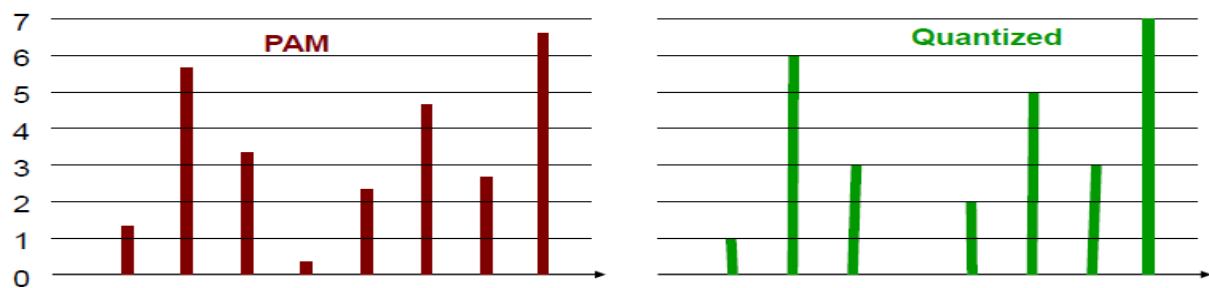


Figure 2.6 Pulse code Modulation technique (Digital Communication Lecture 4:46).

2.5.2 Companding

Companding is the process where the PAM samples are compressed in the transmission side and expanded in the reception side. Companding is the process of compressing the value of PAM samples to fit the non-linear quantizing scale, that result the bandwidth saving. (Shepard 2001:204; Ohrtman 2002:17).

2.5.3 Quantization

The third step is quantization where the values are assigned to each sample in a constrained range. The difference to the original analog signal and constructed digital signal results as quantization noise. Noise reduces voice quality so it is required to reduce the noise. To mitigate this problem we may use more bits per sample, leads to more bandwidth (Ohrtman 2002:17).

2.5.4 Encoding

The last step of PCM is encoding of the signals. This task done by Codec (encoder/decoder), three types of codecs are available they are waveform codecs, source codecs, and hybrid codecs. In waveform codecs make sample and code to the received analog signal without consider as how the signal was generated. Quantized values of the samples are than transmitted to the destination where the original signal is reconstructed. Waveform codecs always ensure with high quality of output. However, the only problem is that it consumes more bandwidth than other codecs. Speech quality degrades if waveform codecs are used low bandwidth (Ohrtman 2002:18).

Source codecs mainly work as all the signals that are coming, match them with a mathematical model. It uses a filter to keep track of the vocal with using voice and unvoiced flag. Afterward, the information transmitted sets a model parameter. At the receiver end, use the same model in reverse to reconstruct the analog signal (Ohrtman 2002:18).

Hybrid codec fills the gaps between the waveform codecs and source codecs. Hybrid codecs are a combination of both techniques which take less bandwidth compared to waveform codecs (Ohrtman 2002:18).

2.6 Transport

Originally, it was very expensive to develop PSTN, so developers were always thinking about how to do it in a cost-effective way. In the beginning, when telephone circuit operation from New York to Los Angeles and the media was copper wire, repeater. Always researchers were thinking about how to use those copper wires to provide maximum efficiency, and the early form of PSTN was TDM. In the 1990s, for long distance users, Interexchange Carriers (IXCs) and for local services users, Local Exchange Carriers (LECs) moved around those transport networks as Asynchronous Transfer Mode (ATM) (Ohrtman 2002:34).

2.6.1 Asynchronous Transfer Mode

Asynchronous Transfer Mode is high speed packet switching which transfers fixed lengths of packet data. It was developed for the high speed transmission of voice, data and video services. ATM switches transfer packets rapidly to other ATM switches or destinations. For this, each ATM maintains a routing table. A routing table refers to the channels to be used for transferring incoming packets and also sets the priority, updated every time when a connection is made or discarded. The ATM works in a flexible way that refers to demands of customers will increase or decrease.

ATM traffic management provides cost efficiency because of effective usage of all circuitry available (Ohrtman 2002:34; Bates 2002:219).

2.6.2 Optical transmission system

At the physical layer, carriers use microwave or fiber optic cable to transport ATM packets which contains the voice and data from switch to switch. Microwave transfer signal energy through one point to another or more point and there should be free line of sight. The node distance from each others can be 25-30 miles with data rate of several hundred's of Mbps. Nowadays with the help of fiber optics microwave are not used so much (Ohrtman 2002:36). However, in the rural area microwave links are still very helpful because it is cost effective. The optical transmission system works as point-to- point connection which utilizes a single optical wavelength which broadcast through an optical fiber. It supports data transmission capacity one million Mbps. Fiber optical lines can be use 1200 km without adding amplifier(Cvijetic 2003:3;Ohrtman 2002:35-36). Transferring optical energy from one point to another point glass or plastic fiber is usually used. At the transmission side uses a light generation (laser), to convert digital information to pulsed light signal uses amplitude modulation technique. After that the light signal can reflect of the side and it is called cladding travel until to the end of the other side. At the receiver side photodetector is used to convert the electromagnetic energy (light) to the original electrical pulses (Cvijetic 2003:3-4; Ohrtman 2002:35-36).

3. OVERVIEW OF SOFTSWITCH

Before Softswitch systems telephone systems have been working based on circuit switching technology. Recently 3G systems utilize packet switched, many value added services are developed during part by part nowadays. Voice over Internet Protocol (VOIP) technology is one example. Different types of services are provided through gatekeepers, Session Initiation Protocol (SIP), proxy servers and call agents. Those are developed currently which can control calls name as Softswitch. Softswitch controls different types of devices such gateways, media and signaling gateways. Softswitch consists of 4 different planes in architecture as shown in Figure 3.1 (Guizani 2004:361-362; Xu et al. 2003: 2744; Lemay et al. 2006:3299-3300).

- Transportation plane.
- Call control and signaling plane.
- Service application plane.
- Management plane.

3.1 Transportation plane

Transportation plane transports data packet through the help of IP network which consists of call control signal and media data. Different types of gateways such as access gateway, signaling gateway, media gateway and IP terminals are presented in that transportation plane. Transportation plane consists of three parts which are (Guizani 2004:361-362; Xu et al. 2003: 2744; Lemay et al. 2006:3299-3300).

- I. IP part, different types of switch and routers are presented to setup connection to IP backbone.

- II. Interworking part, which contains different types of gateways to establish the connection of different types of network through the help of Trunking Gateways (TG), Signaling Gateways (SG) such as PSTN and SS7 network.
- III. None IP part, connected with none IP terminal and mobile network such as access gateways connected to none IP Terminals or phone.

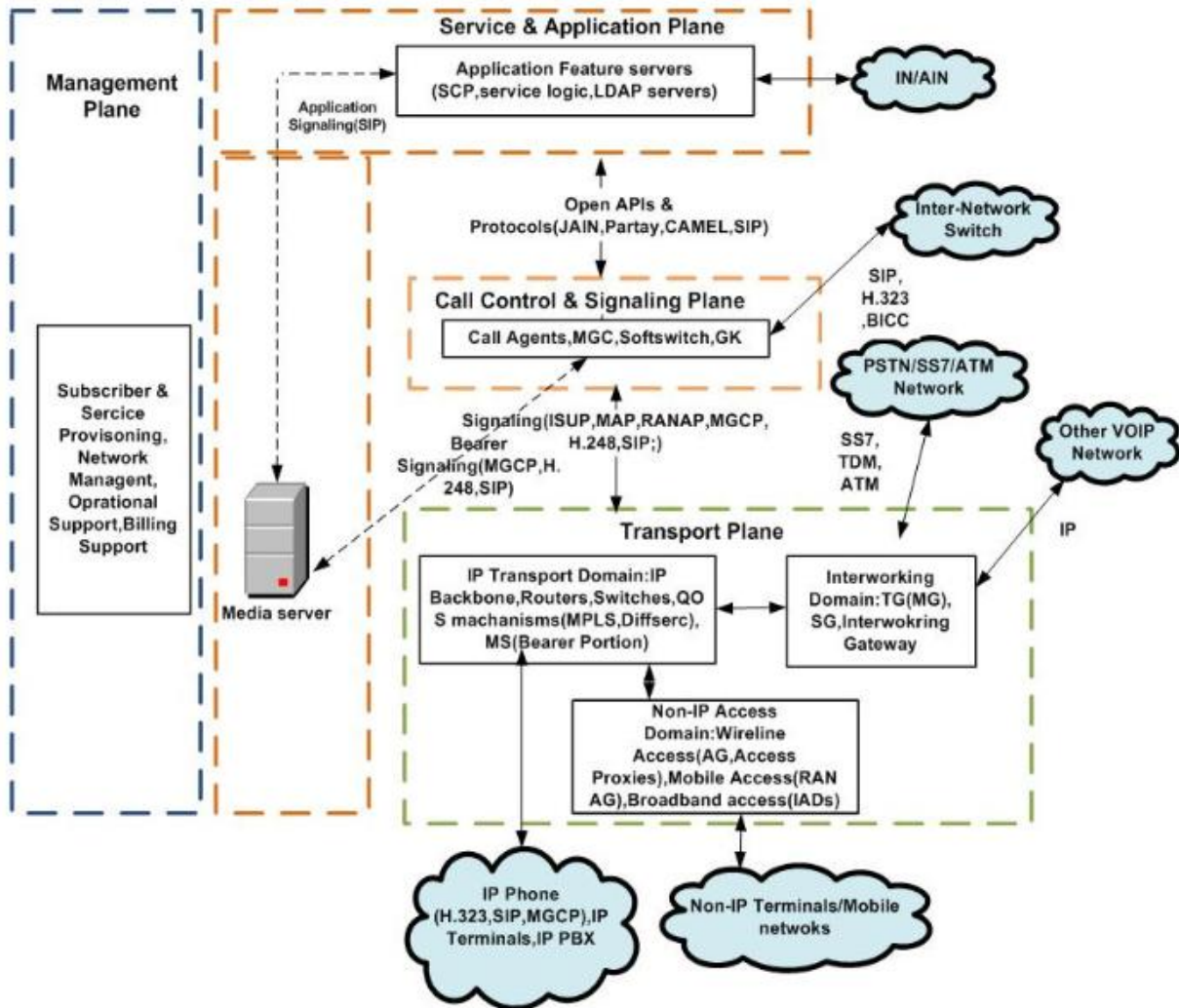


Figure 3.1 Architecture of Softswitch (Softswitch multicriteria analysis for software quality based on international packet communication Consortium (IPCC) Reference architecture 2006:3300).

3.2 Call Control and signaling plane

This is the core part of Softswitch. The main responsibilities of this plane are call controlling and routing of the received call from transportation plane. Signaling protocols contains also traditional circuit switching network protocols, such as ISUP, is used for SS7 signaling. It also contains Mobile Application Part (MAP) as well as VOIP protocols, such as Session Initiation Protocol (SIP), H323 protocol, Media Gateway Control Protocol (MGCP)(Guizani 2004:361-362; Xu et al. 2003: 2744; Lemay et al. 2006:3299-3300).

3.3 Service and application plane

Different types of application servers are presented in the service and application plane. The main task is to create a Service Creation Environment (SCE) and Service Logic Execution Environment (SLEE). SCE is a graphical user interface where the user can develop the system easily. After that the Service logic Execution Environment (SLEE) compiles the program and established communication to the control and signaling plane through the Application Programming Interface (API). This plane also include media server which can perform conferencing and voice mail services (Guizani 2004:361-362; Xu et el. 2003: 2744; Lemay et el. 2006:3299-3300).

3.4 Management plane

Management plane supervised the above three planes. Moreover, it performs other tasks such as billing and network management and operational support. This plane has not been presented as in most softswitch implementation. It differs from one vendor to another (Guizani 2004:361-362; Xu et el. 2003: 2744; Lemay et el. 2006:3299-3300).

3.5 Softswitch based wireless networks architecture

In Softswitch based wireless network, Softswitch works as a complete feature of Mobile Switching Center (MSC) server. The following Figure shows softswitch based GSM (Global System for Mobile Communication) networks.

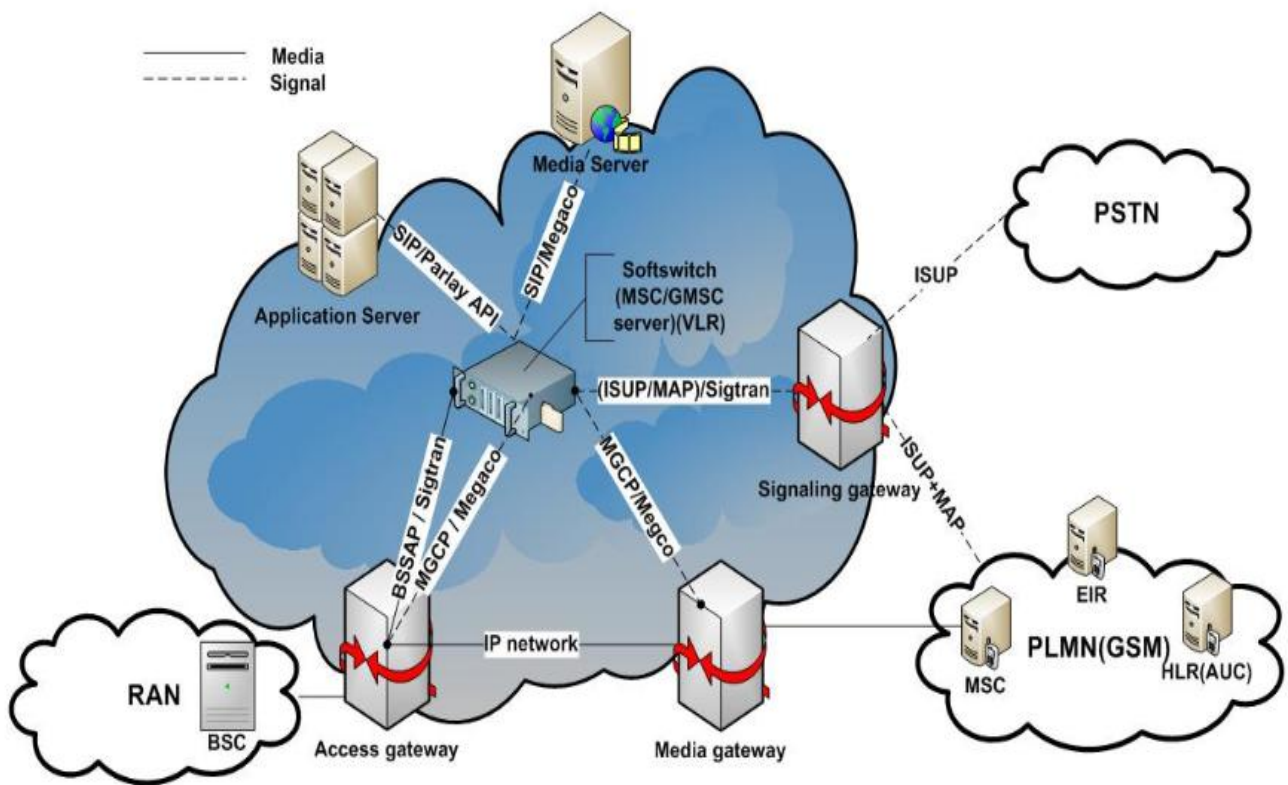


Figure 3.2 Softswitch based wireless networks architecture. (Research on service solutions in Softswitch based wireless networks 2003:2745).

To connect Softswitch through Base Station Controller (BSC) via Access Gateway (AG), softswitch support Base Station System Application Part (BSSAP) over SIGTRAN. Internet Engineering Task Force (IETF) made possible the signaling operation via

Signaling Transport (SIGTRAN) protocols. With the purpose, to allows reaching any subscriber in one network to other network (Hemant, Dantu, Wijesekera & Jajodia 2006:32). Figure 3.2 shows that Softswitch can construct interconnection to the Public Land Mobile Network (PLMN) through Signaling Gateway (SG) and Media Gateway (MG). Signaling gateway transfers signal (ISUP, TUP, TCA, INAP, and CAP) using different type's interfaces through SS7 networks and IP networks. Signaling Gateway (SG) transfers signal between circuit switched and IP network. SG can terminate SS7 signaling over an IP network to Media Gateway (MG) or can be another Signaling Gateway (SG). Media gateway works as a media for establish connection between circuit switch network (PSTN, PLMN) and IP networks. Softswitch used Media Gateway Control Protocol (MGCP/Megaco) for controlling operation through to media gateway. MGCP/Megaco call processing operation is addressed in greater details in chapter 4 (Alja & Brodnik 2004:159-160; Xu et al. 2003: 2744-2745).

3.5.1 Softswitch based call setup between two softphone

To established a simple call to IP user, a device used that is located close to the Softswitch. Which works as a translation device (H223, Gatekeeper, and Sip Proxy) can easily identify user number of IP address. After complete the identify procedure call can be easily established. (Alja, Brodnik 2004:159-160). Fig 3.3 shows the SIP based softswitch connection between Softphone 1 and Softphone 2. In this situation softswitch act as a SIP proxy server and also process the call. The task for SIP proxy server is to modify and forwarded the request. SIP based softswitch end point recognized as User Agent (UA), such as SIP phones, softphones and telephone gateways. Where softswitch is intermediate network elements between the end points and engage them by routing of SIP message based on a logical SIP address. Softswitch also executes function of authentication, authorization and signal compression (LI, Su & Yang 2003:66-67; Hyun-woo, Jinsul, Won & Byung-sun 2007:2960-2961). "A logical SIP URI address consists of a domain and identifies a UA ID number. The UAs belonging to a particular domain register their

locations with the SIP Registrar of that domain by means of a REGISTER message. The Registrar saves the location information of that particular UA in a location database. Softswitch refers to that location database for address resolution between logical SIP URI destinations and physical locations at IP addresses” (Hyun-woo et al. 2007:2961).

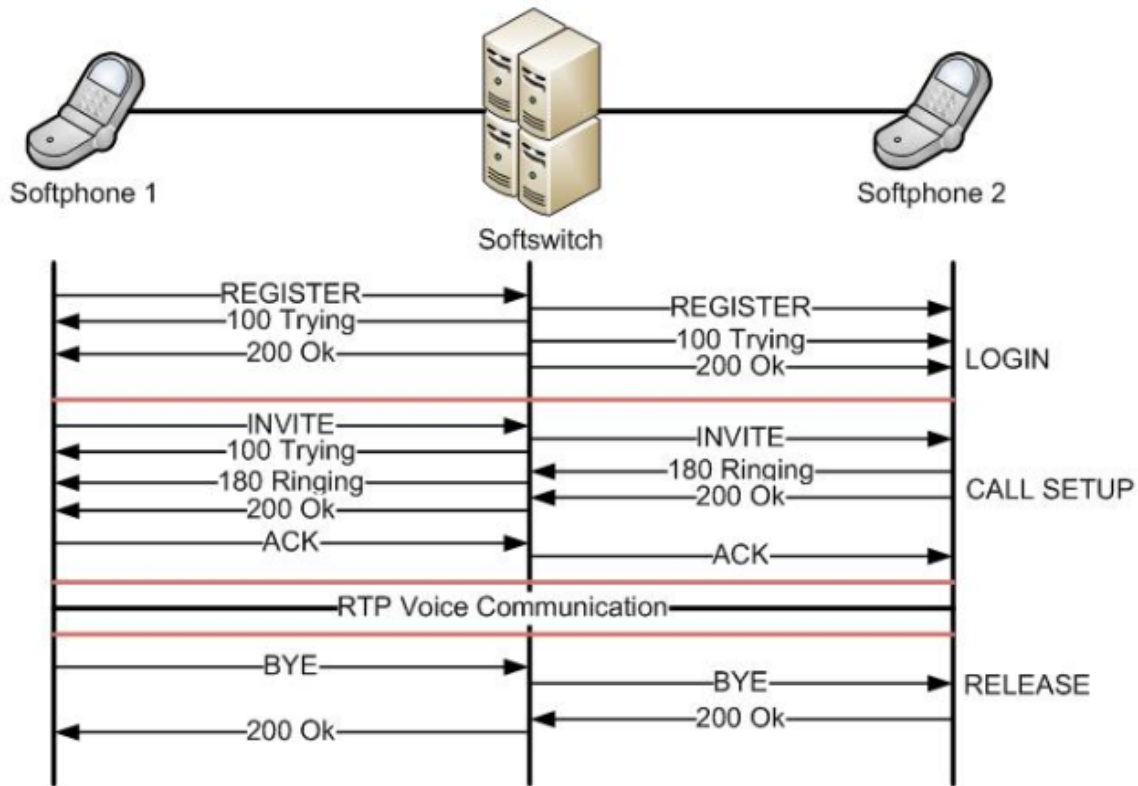


Figure 3.3 Procedures of call setup and release between Softswitch and Softphone (A Study of the Real-time Message Report Procedures and Management Schemes for the Quality Guaranteed VoIP Services 2007:2961).

If any IP user wants to call other PSTN users then connection can be made easily through the help of media gateway (Alja & Brodnik 2004:159-160).

3.6 Softswitch based wireless network services

Softswitch based wireless network services can be described in three ways they are as follows:

- a) Percepts the existent services
- b) Development of Softswitch services
- c) Overview of application servers services

3.6.1 Percepts the existent services

Short Message Service (SMS) was developed before in cellular network but softswitch takes over these services and provide some value added function. SMS is well known text based service with lowest rate. Softswitch system can handle this service. Figure 3.4 shows how SMS works in Softswitch system (Xu et al. 2003: 2744-2745).

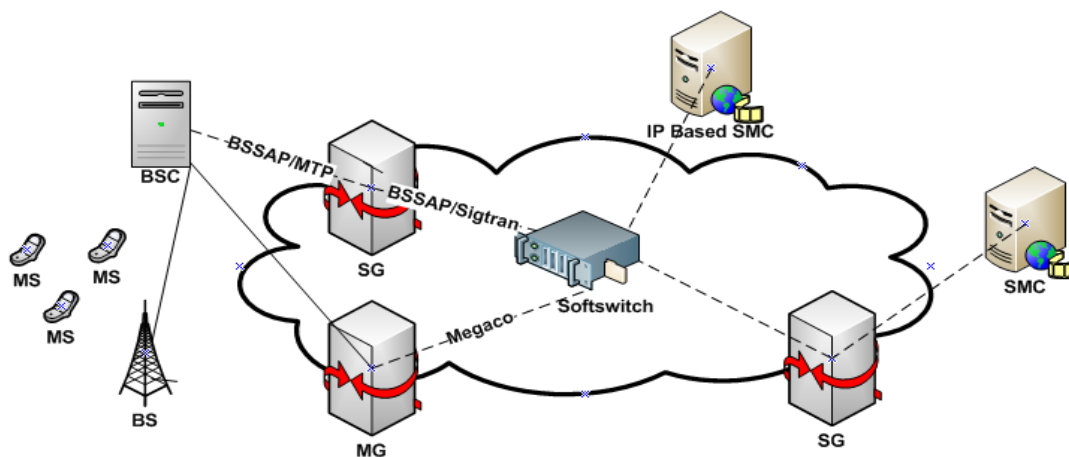


Figure 3.4 SMS in softswitch system (Research on service solutions in Softswitch based wireless networks 2003:2745)

With the improvement of SMS, Service Message Center (SMC) has begun to transfer to IP networks. As an IP based structure softswitch system could better collaborate with IP based SMC by supporting SS7 over IP technologies such as IETF Sigtran. To provide powerful services Mobile Intelligent Network (MIN) plays great roles for establishment PLMN services. In the architecture of CAMEL, Mobile Service Control Point (MSCP) provides Service Control Function (SCF), MSC provides Service Switching Function (SSF) and Intelligent Peripheral (IP) provides Special Resource Function (SRF). In setup connection between softswitch and MIN system softswitch can easily connect with MSCP through Signaling Gateway (SG) and also connect IP via Media Gateway (MG). Figure 3.5 shows interoperation between softswitch and MIN (Xu et al. 2003: 2744-2745).

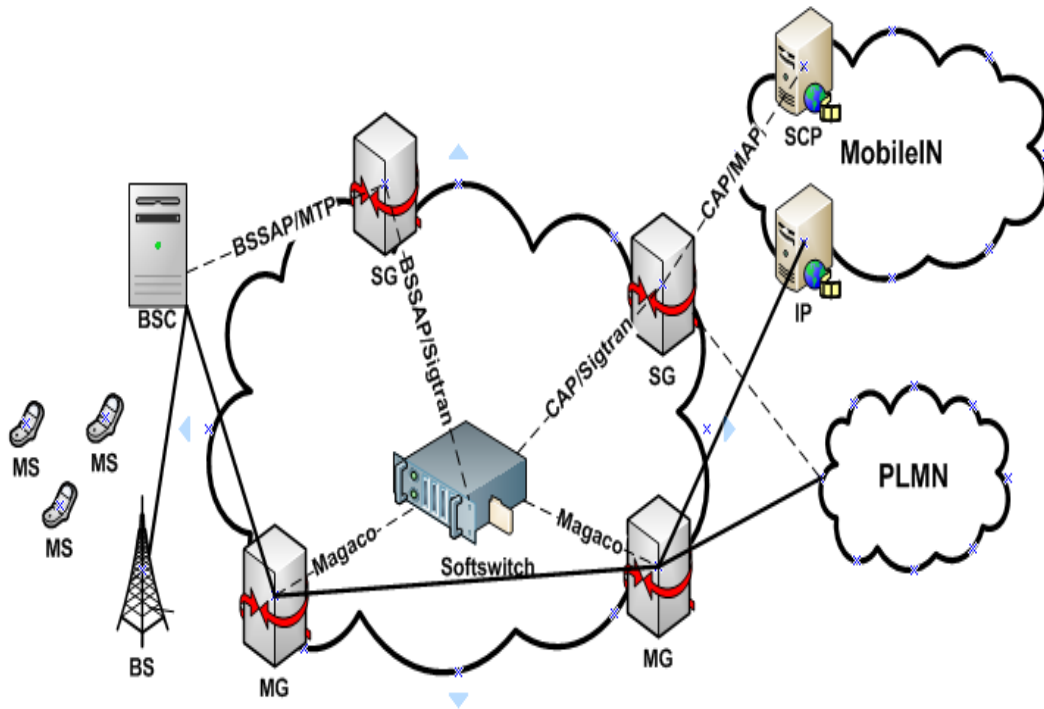


Figure 3.5 Interoperation between Softswitch and MIN (Research on service solutions in Softswitch based wireless networks 2003:2746).

3.6.2 Development softswitch services

We know from softswitch architecture that softswitch support different types of signaling protocols, such as ISUP, SIP, H.323, MGCP, MAP and BSSAP etc. The entire protocols work for call control operation. Those signaling protocols are used to control different types of peripheral devices to receive data or messages and send them after processing (Xu et al. 2003: 2744-2745).

3.6.3 Overview of application server's services

Application Servers (AS) executes the services and provides the best control capabilities of softswitch. Application servers has to receive into account is just how to construct the best call control capabilities. AS unaware which entities actually provide certain capabilities. Furthermore, services deployed in AS can be easily shared by all subscribers connected to softswitch system. The service interface between AS and softswitch can be SIP and Application Programming Interface (API) based. Figure 3.6 describes the application server based service environment (Xu et al. 2003: 2744-2745).

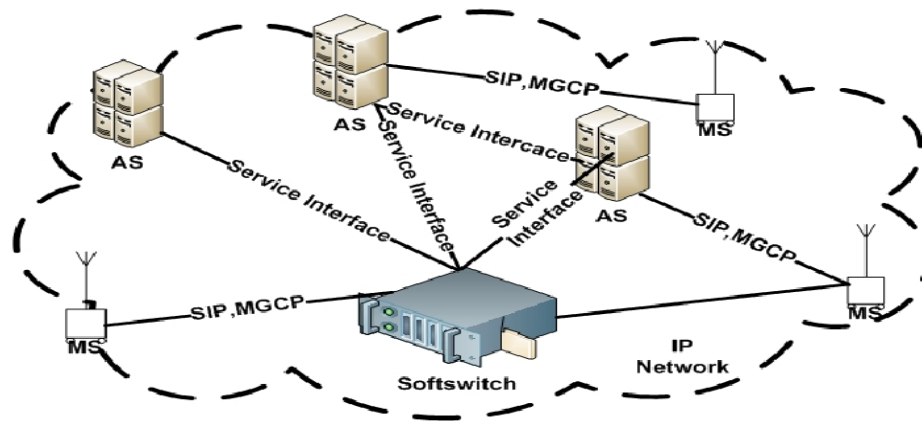


Figure 3.6 Application Server Based Service Environment (Research on service solutions in Softswitch based wireless networks 2003:2746).

3.6.3.1 SIP based interface service

In SIP based interface service when softswitch receives any request for service, it processes the user request for authentication and authorization. If it accepts the authorization, then softswitch accepts the request from user as a SIP message and send it to the application server. Application server processes the request with executing certain logic and returns corresponding messages if needed. After that softswitch will accept and return those messages to the format that subscribers terminals could understand. Then send the revised messages to the subscriber (Xu et al. 2003: 2745).

3.6.3.2 API based interface service

API based advance patterns develop by service API for service improvement. Service developers try to give assurance of integrity and security of telecommunication networks. While utilize some basic service capabilities with service API. The recognition of service API is increasing significantly by giving facilities like convenient service development manners and powerful service capabilities. At present some organizations identify numerous service APIs such as Parlay API, OSA API and many mores. To transport essential information between Softswitch and AS is the key difficulty of service APIs. The problem can be solved in two ways (Xu et al. 2003: 2745-2746).

- SIP based transportation mainly extent SIP and for transportation encapsulate required information into SIP messages.
- Common Object Request Broker Architecture (CORBA) based transportation implies to accept a CORBA platform under softswitch and AS.

3.7 Functional elements of Softswitch

Functional elements have special role in the entire system, those are logical elements. Those elements present different planes in the same time. Figure 3.7 consists of 12 different functional elements those functions describe the ISC reference architecture important parts. In the following this components are describe respectively (Lemay et el. 2006:3300).

3.7.1 Application Server Function (AS-F)

In this section services are executed and provide application to the system. AS-F are as follows (Lemay et el. 2006:3300; Ohrtman 2002:315-316).

- AS-F can request to the MGC-F to terminate the calls such as voice mail and conference services.
- AS-F can control MS-F like media handling function.
- AS-F can also link web application and can use API for create a service.
- AS-F can be use another AS-F for operate and execution additional or complicated services.
- To work together AS-F and MGC-F can provide some extra services such as 3 ways calling and call waiting. Those can be done without use any protocol between AS-F and MGC-F. The Vendors usually uses API when the implementation can be done in same system.

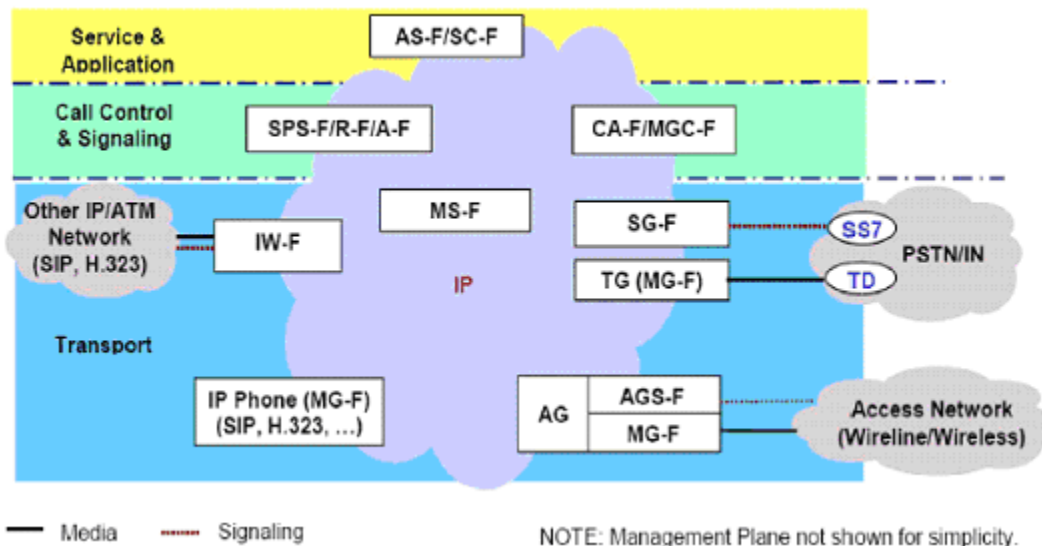


Figure 3.7 Functional elements of Softswitch ISC reference architecture (Softswitch 2002:312).

3.7.2 Service Control Function (SC-F)

The SC-F exists when AS-F control the service logic of a function. It uses different types of protocol such as Intelligent Network Application Part (INAP), Common Alerting Protocol (CAP), Mobile Application Part (MAP) (Lemay et al. 2006:3300; Ohrtman 2002:316).

3.7.3 Call Agent Function (CA-F) and Interworking Function (IW-F)

CA-F and IW-F works together when MGC-F exists. MGC-F takes care of the call control operation in the presents of CA-F. There are different types of protocols and APIs are working in a CA-F such as SIP, H.323, MAP, BSSAP, ISUP and TCAP. APIs are Parley and JAIN. When MGC-F is handles signaling to setup connection between different signaling network such as SS7 and SIP. This process is completed with the help of IW-F.

Different types of IW-F protocols are interoperate such as H.323/SIP (Lemay et al. 2006:3300; Ohrtman 2002:313).

3.7.4 Media Gateway Controller Function (MGC-F)

MGC-F can control the call and signaling operation to one or more media gateways. MGC-F is as follows (Lemay et al. 2006:3300; Ohrtman 2002:312-313).

- MGC-F looks after the bearer interface to the MG-F and exchange message between two MG-F, for example IP Phones.
- MGC-F controls every call through media gateway.
- Signaling information can terminates from other endpoints and external network.
- MGC-F can work together with AS-F to provide the efficient services to the subscribers.
- MGC-F can handle the bandwidth and MG-F port management. Protocols are MGCP and H.248.

3.7.5 Call Routing Function (R-F), Accounting Function (A-F) and SIP Proxy Server Function (SPS-F)

Call R-F work to find out the route path for setup connection and that information send to the MGC-F. Where Call A-F is works as to collect account information for billing related tasks. A-F generated details bills for every session. R-F protocols are ENUM and TRIP. A-F protocols are RADIUS and AuC. SIP Proxy server Function (SPS-F) can work as combination of Call Routing Function (R-F) and Accounting Function (A-F) (Lemay et al. 2006:3300; Ohrtman 2002:313-314).

3.7.6 Media Server Function (MS-F)

MS-F main task is to take care of the server. This especially can handle the AS-F and MGC-F, to facilitate the media processing in the packetized media streams. MS-Fs tasks are as follows (Lemay et al. 2006:3300; Ohrtman 2002:317-318).

- MS-F support multiples codecs and control multiples AS-Fs and MGC-Fs.
- MS-F can perform multiples task concurrently such as:
 - Detection of digit
 - Can recognize the speech
 - Can record Multimedia stream
 - Can also create speech from text
- Protocols are SIP, H.248, and MGCP

3.7.7 Signaling Gateway Function (SG-F) and Access Gateway Signaling Function (AGS-F)

SG-F exchange signaling information through gateway between different types of network such as VOIP and PSTN network (SS7/TDM or BICC/ATM) based. For wireless based infrastructure SG-F provides signaling information through gateway between IP and PLMN networks (SS7/TDM or BICC/ATM) based. AGS-F exchange signaling information through gateway between different types of network such as VOIP and Circuit switch access network (V5 or ISDN) based. For wireless based infrastructure AGS-F provides signaling information through gateway between IP and PLMN networks (TDM or ATM) based (Lemay et al. 2006:3300; Ohrtman 2002:314-315).

3.7.8 Media Gateway Function (MG-F)

MG-F provides the gateway between the IP and circuit switch network. For an example is IP and PSTN network, or two packet network such as IP and 3G. MG-F primary role is to transform media from one transmission format to another for example between circuits and packets, between ATM packets and IP packets. MG-Fs are as follows (Lemay et al. 2006:3300; Ohrtman 2002:315-317).

- It process media processing function such as media packetization, buffer and jitter management.
- It can generate call progress tone, Dual Tone Multi Frequency (DTMF) generation.
- It detects signaling and media such as DTMF detection, voice activity detection.

4. SOFTSWITCH SIGNALING

Using IP based infrastructure provides superior facilities to distinguish the voice traffic Real Time Protocol (RTP) from signaling (H.323, SIP, and MGCP). The major role of softswitch is to handle the signaling where as in the voice part RTP follow voice across through voice core network between VOIP terminals or media gateways. Figure 4.1 shows a simplified model of softswitch (Zelenika & Magdic 2007:317-318).

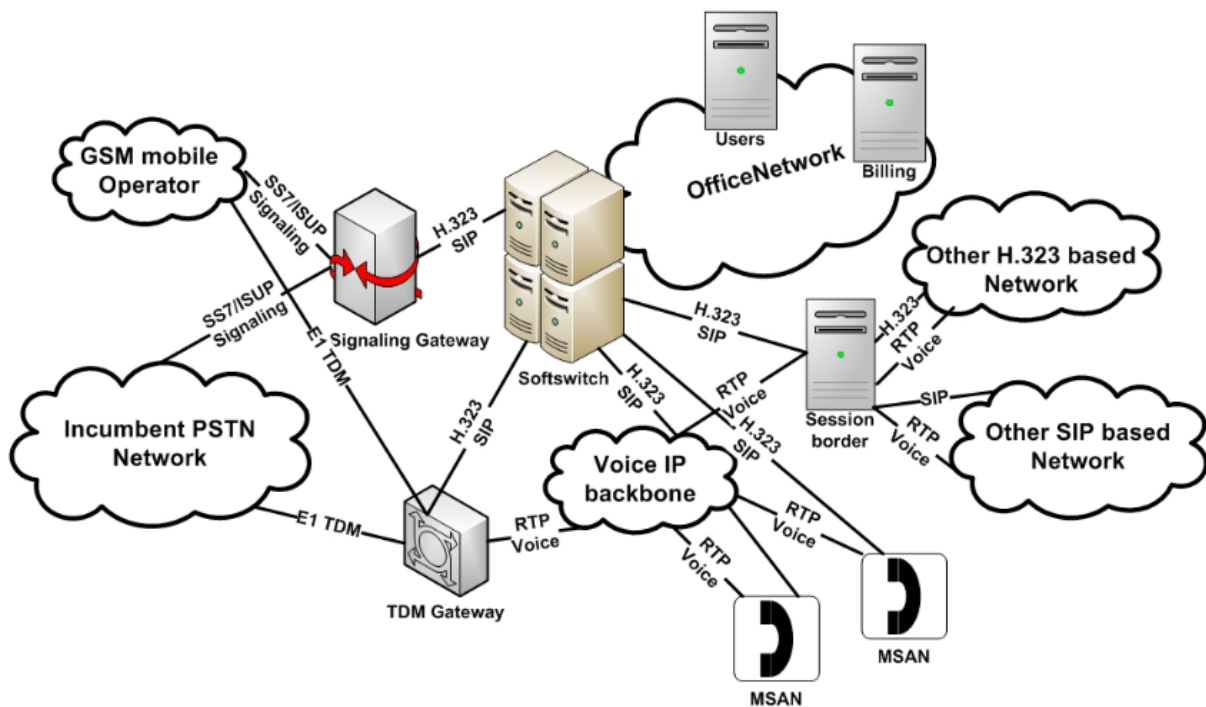


Figure 4.1 Simplified Model of Softswitch (Network Building Block of Fixed NGN Telecom Operator 2007:318).

Figure 4.1 shows that two different types of network can be recognized through a Media Gateway (MGW) that converts the PCM encoded voice to RTP based IP voice. Signaling gateway converts the as usual signaling system 7 (SS7) into H.323 or SIP signaling protocol which can be identified by the core softswitch (Zelenika & Magdic 2007:317-318).

4.1 Signaling System 7 (SS7)

Nowadays SS7 is the main part in the telecommunication system for call control signaling part in many of the Telecommunication Companies. SS7 provides fast and efficiency services to exchange signaling information between switches. To setup call between two users the network resources have to be set. Because of exchanging information for call management and the information should be in message form. SS7 work as signaling service for data network for PSTN, ISDN, and GSM. In SS7 signaling message can transmit very high speed. Moreover, in call progress time signaling information can be exchanged (Stafford 2004:55-56; Prasad 2004:394-395; International Engineering Consortium white paper).

4.2 SS7 Protocol Layers

SS7 protocol layers are describing in Figure 4.2.

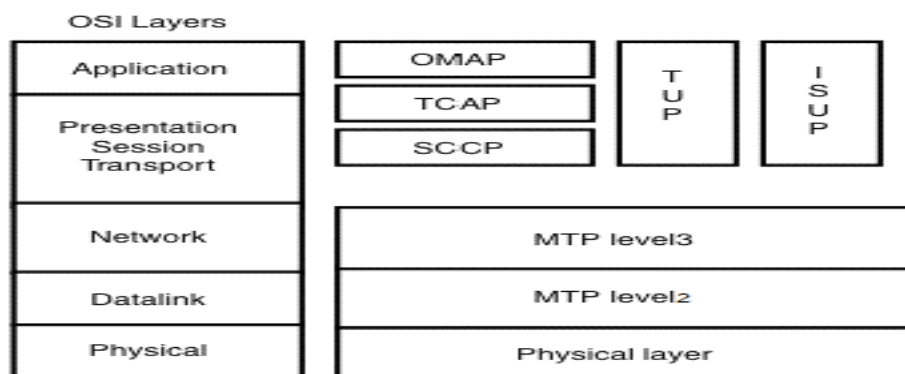


Figure 4.2 SS7 Protocol layers (Principles of Digital Communication Systems and Computer Networks 2004:400).

4.2.1 Physical layer and Message Transfer Part Level 2 (MTP2)

The physical layer transmit row signaling data at 56 or 64 kbps. MTP2 works as error detection and correction similar to the data link layer. MTP2 maintains a sequence number but this number does not provide any benefits. It is differ from TCP because TCP maintain sequence number and it has something to perform in the protocol stack such as recall the TCP in the transport layer (Prasad 2004:400-401; Stafford 2004:56-57; International Engineering Consortium white paper).

4.2.2 Message Transfer Part Level 3 (MTP3)

MTP2 and MTP3 both of them together called MTP. This part handles network management such as addressing, routing, alternative routing and congestion control. Its message handling done in following ways (Prasad 2004:401; Stafford 2004:57; International Engineering Consortium white paper).

- Message discrimination. It checks if the current node is the destination if it is, then message to be send to the distribution function otherwise send to the message routing function.
- Message routing. MTP3 provides point to point routing and it has limited routing intelligence.
- Message distribution. Its passes the message to the higher layer.

4.2.3 Signaling Connection Control Part (SCCP)

For increasing the quality of service, SS7 needs to query the database. MTP cannot carry all those operation they have some subsystem such as toll free number, repeat dialing,

advanced intelligent network and call processing. SCCP provides the facility to addresses application within a signaling point. SCCP can provide routing end to end point to access the database. SCCP cannot perform the database quires it only confirms that those quires reach their destination (Prasad 2004:401; Stafford 2004:56-57; Report of Ulticom service essential solutions).

4.2.4 Transaction Capabilities Application Part (TCAP)

TCAP message must be delivered to individual applications with the node they address, these message use SCCP for transport. Whenever SS7 needs a database query TCAP takes part in action. The query and answer formulated as TCAP message. Some extra services take care through TCAP for example calling card services. Some of the TCAP applications are (Prasad 2004:401-402; Stafford 2004:56-57; Report of Ulticom service essential solutions).

- AIN: Advanced intelligent Network
- ANSI 41: It is known as T1A or E1A, wireless intersystem operation used in North America and also some part of Asia.
- GSM MAP: Global System for Mobile Communication (GSM) Mobile Application Part (MAP) is used to connect to wireless communication system. Wireless network maintain each user information and location in a database called home location register.
- CAP: Customized Application for Mobile Network Enhanced Logic (CAMEL) Application Part provides intelligent network capabilities for GSM.

4.2.5 Operation Maintenance and Administration Part (OMAP)

“OMAP defines messages and protocols to administer SS7 networks. The functions include validation of route tables and diagnosis of links. OMAP is an application that uses the TCAP and SCCP services.”(Prasad 2004:402).

4.2.6 Telephone User Part (TUP) and ISDN User Part (ISUP)

TUP is working as signaling backbone of switching elements to establish or release calls from circuit switch network. The ISUP is used to set up telephone call in public switch network that control the signaling. An ISUP supports voice and data service, but does not support the broadband services. For broadband support another new technology developed which know as Broadband ISUP (BISUP). ISUP works both analog and digital circuit. This is the reason to replace the TUP, because TUP does not support data transmission and digital circuit. Now ISUP is also used in wireless communication for establishing trunk connection between switching centers. ISUP provides two types of services basic and supplementary services. Basic service is used to setup connection in the circuit network which is known as voice and data service. Supplementary service is known as other circuit oriented service. That service occurs after call paths are established. Figure 4.2 describe the simple call flow structure for ISUP (Prasad 2004:401; Stafford 2004:58; Russell 2000:355-357).

In this case, the originating switch connects with destination switch with direct bearer connection. After the originating switch receives the dial digit it selects a channel and allocates for this call (Stafford 2004:58-59).

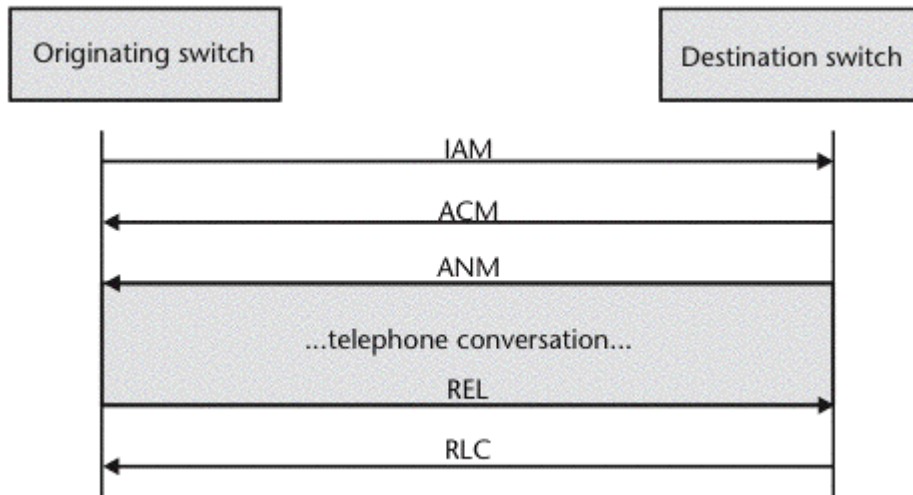


Figure 4.2 Simple ISUP call flow diagram (Signaling and Switching for Packet Telephony 2004:59).

After that sends Initial Address Message (IAM) to the destination switch. After the destination switch receives the IAM then it sends the Address Complete Message (ACM) to the originating switch, and send ring to the called subscriber. After that when the subscriber receives the call then destination switch sends another message which is know as Answer Message (ANM).The billing is done in the originating switch part. When the conversion will finish then the originating switch sends Release Message (REL) to the destination switch. After that the destination switch receives the REL its sends Release Complete message (RLC) to the originating switch (Stafford 2004:58-59).

4.3 Signaling Protocol and Interface

In Mobile Softswitch Solution (MSS) to establish circuit switch connection the Bearer Independent Call Control Protocol (BICC), Gateway Control Protocol (GCP), IP Bearer Control Protocol (IPBCP) and NbUP protocol are used. Figure 4.3 shows signaling protocols that are used in the layered architecture model (Ericsson AB 2006:43).

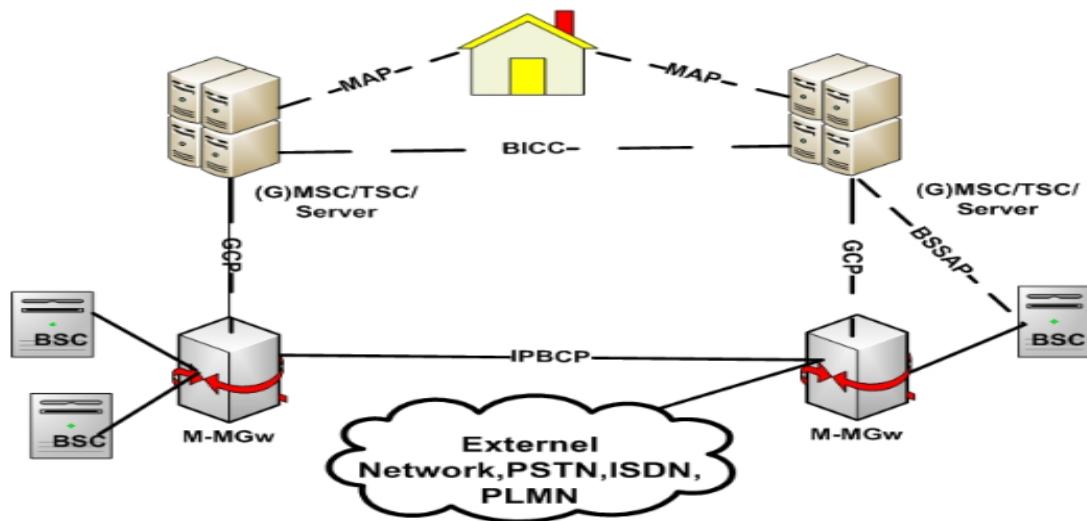


Figure 4.2 Signaling protocol (GSM mobile softswitch solution R3.1 introduction 2006:43).

4.3.1 Bearer Independent Call Control (BICC)

Day by day the use of internet subscribers are increasing then the issue is to provide effective transmission. The earlier ISUP signaling is only adjustable to TDM. It is not sufficient for packet based voice and circuit based video data for IP or ATM network. To rise above the limitation BICC can be used a good solution. BICC was developed by ITU-T. ITU-T point of view, on this was that BICC protocol was 100% compatible like ISUP .Which ensure that two different PSTN customers can connect each other smoothly via IP network. BICC used to control the server such as MSC and GMSC server. BICC handles voice and data services, it runs in both IP and ATM (Jeong-Je & Nak-Po 2008:699; Bannister, Mather & Coope 2004:536).

4.3.2 Supported Services by BICC

All services that ISUP can support BICC can also support. The services are basic service, fax, 64 kbps, speech 3.1 kHz audio, access delivery information, suspend and resume and so on. Other addition services are as Caller Line Identification Presentation (CLIP), Call forwarding (Ericsson AB 2006:47; ITU-T Q.1902.1 2001:13).

4.3.3 Types of BICC

ITU developed two types of BICC, they are describe as follows

- BICC capability Set 1(CS1).
- BICC capability set 2(CS2).

In BICC capability set 1 (CS1) only ATM transport available so, only the TDM network operator can transfer their system in ATM. It was mainly developed for packet based network. The disadvantage of that CS1 is, it always assumes a network model but it is true that there is no any physical separation between the call control and bearer control nodes. Depending of that reasons it is also suspected that there is one control server for every media gateway which function incorporated in one node. For resolve this problem BICC capability set 2(CS2) developed by ITU (Bannister et el. 2004:536; Ericsson AB 2006:48).

In BICC capability set 2(CS2) can split the physical layer in call control and bearer control. It supports the M-MGws selection, allow many to many relationship. That can help to set relationship between the control servers and M-MGws. Some other supplementary services supported that ISUP can do, for example call forwarding on busy (Bannister et el. 2004:537; Ericsson AB 2006:48).

4.3.4 Forward and Backward Bearer setup

In forward bearer setup is the same direction as Initial Address Message (IAM) message. If the bearer established in the reverse direction (IAM message), then it is called backward bearer. In Figure 4.3 describe the establishment of forward and backward bearer.

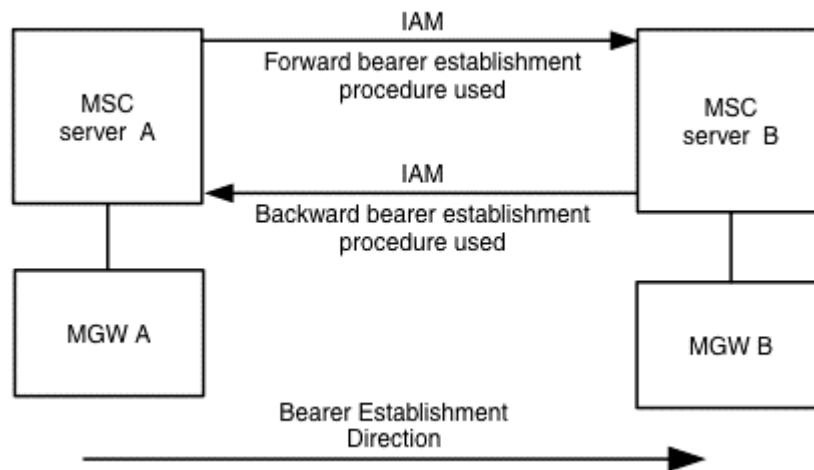


Figure 4.3 Forward and Backward bearer establishment (Convergence Technologies for 3G Networks: IP, UMTS, EGPRS and ATM 2004:538).

MSC server admission calls if there enough bandwidth is available. MSC server A create IAM message for forward bearer establishment. And for backward bearer establishment MSC server B create IAM message (Bannister et al. 2004:538; Ericsson AB 2006:50-51).

4.3.5 BICC messages and parameters

The basic call setup procedure is shown next in Figure 4.4 where Call Service Function Node (CSF-N) and Bearer Control Function Node (BCF-N) are used. The process starts with Initial Address Message (IAM) which is used for initiate the call setup. This message contains information of called party number. It is forward the message to the destination

CSF-N who will serve the called party. Otherwise, it can be forward to the PSTN if the CSF-N connected to the PSTN. Application Transport Message (APM) used to forward the non BICC information between the BICC users. However, in the presence of IAM, the APM works as reverse direction (Bannister et al. 2004:538-540).

The APM contains bearer information of remote gateways address such as IP address and RTP port number. After receiving the APM the CSF-N says it is local BCF-N to setup bearer with remote BCF-N (this request carried H.248 transaction). After that when the call routing has done on the way to destination. Afterward, the Address Complete Message (ACM) sends to reverse direction which indicated that all addressing information has been received. After that the destination phone ringing and the ringing tone also played to the caller party phone. Once the called party received the phone then the Answer Message (ANM) sends to the reverse direction. Afterward, Ringing tone signal removed from both parties line and media is connected both parties direction for voice conversation. As soon as any of the users hang up the phones then Release Message (REL) sends after that Clear Message (RLC) used to clear the call (Bannister et al. 2004:538-540).

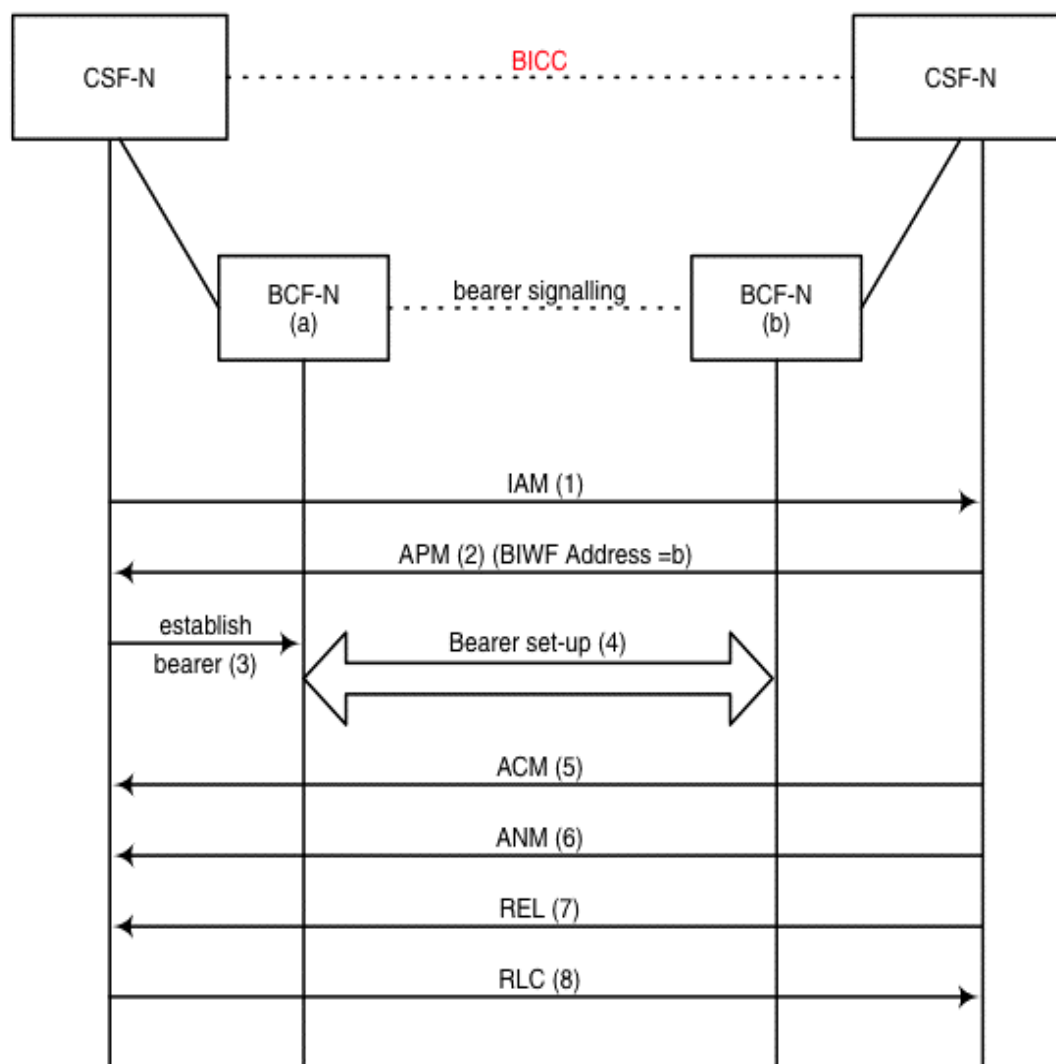


Figure 4.4 BICC forward bearer establishment (Convergence Technologies for 3G Networks: IP, UMTS, EGPRS and ATM 2004:539).

4.3.6 Codec Negotiation

Codec negotiation is a process of established common codec between the end user. In the bearer setup procedure the M-MGW provides a list of available codecs. The priority can be set their in order to the highest priority. The codec is selected before forwarding the IAM

message. The selected codec is returned in the BICC APM message after that the selected codec is to be used for communication. In Figure 4.5 describe the voice compression process in GSM core network (Bannister et al. 2004:547; Ericsson AB 2006:23-24).

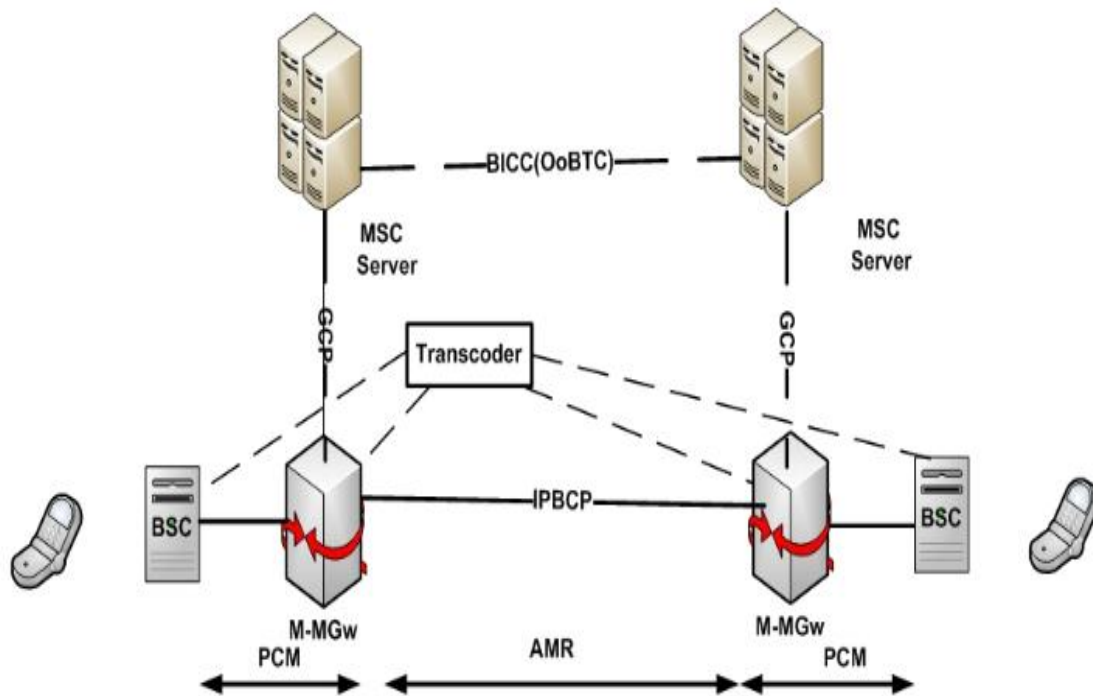


Figure 4.5 Voice Compression in GSM core network part 1 (GSM mobile softswitch solution R3.1 introduction 2006:24).

The MSC server uses Out-of-band Transcoder control in BICC CS2 signaling system to select the best optimal codec type. The OoBTC in MSC server take care the call control plane, and the M-MGW takes care the user plane. In other case to connect to another external network such as PSTN and other PLMN to IP network, Adaptive Multi Rate (AMR) coded used to connect between M-MGW. In Figure 4.6 describe the voice compression procedure between the IP and External network (Bannister et al. 2004:547; Ericsson AB 2006:23-24).

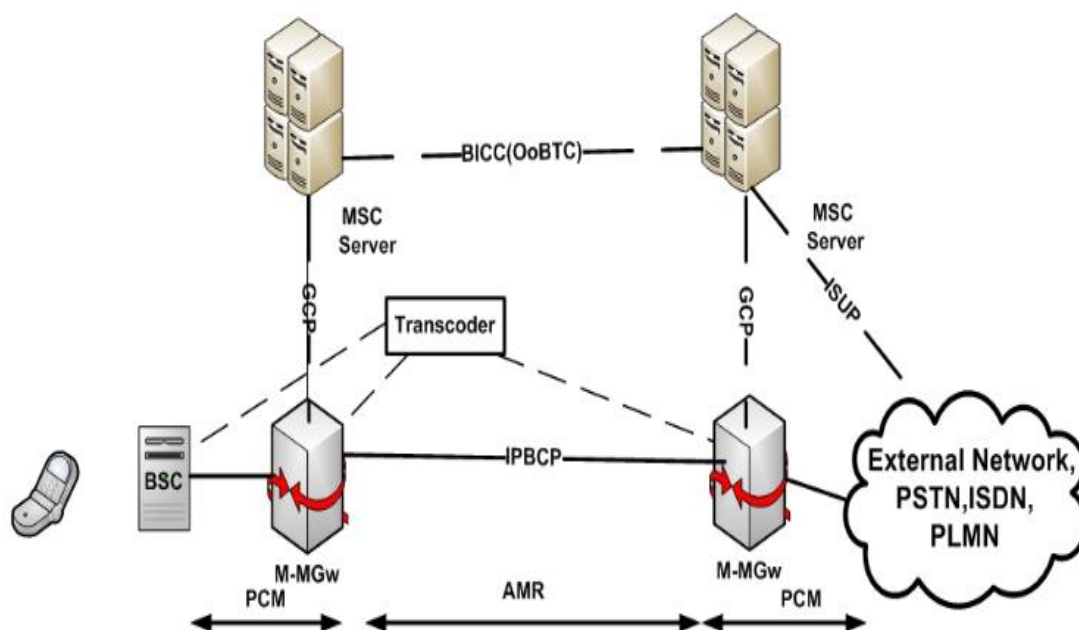


Figure 4.6 Voice Compression in GSM core network part 2 (GSM mobile softswitch solution R3.1 introduction 2006:24).

4.3.7 Notification of Bearer Setup

After bearer establishment some signaling process need to be notified, before sending call control messages. For an example ISUP Continuity messages (COT).For this reason MSC server notifies the bearer setup confirmation at the call control level. This process can be done by sending an acknowledgement through Application Transport Message (APM). The benefits of using notification of bearer setup is inter-operability reasons (Ericsson AB 2006:52).

4.3.8 IP Bearer Control Protocol

IPBCP permits to carry media information through IP network between two M-MGw. IPBCP uses media stream information such as RTP port number and IP address to established IP bearers. IPBCP has established a bearer on per call basis in Softswitch. IPBCP has designed in such a way that works as tunnelling protocol in the vertical end Getaway Control Protocol (GCP). The horizontal side BICC protocol uses for communicate bearers. Those can be shown in the Figure 4.7 (Bannister et el. 2004:542-543; Ericsson AB 2006:67-68).

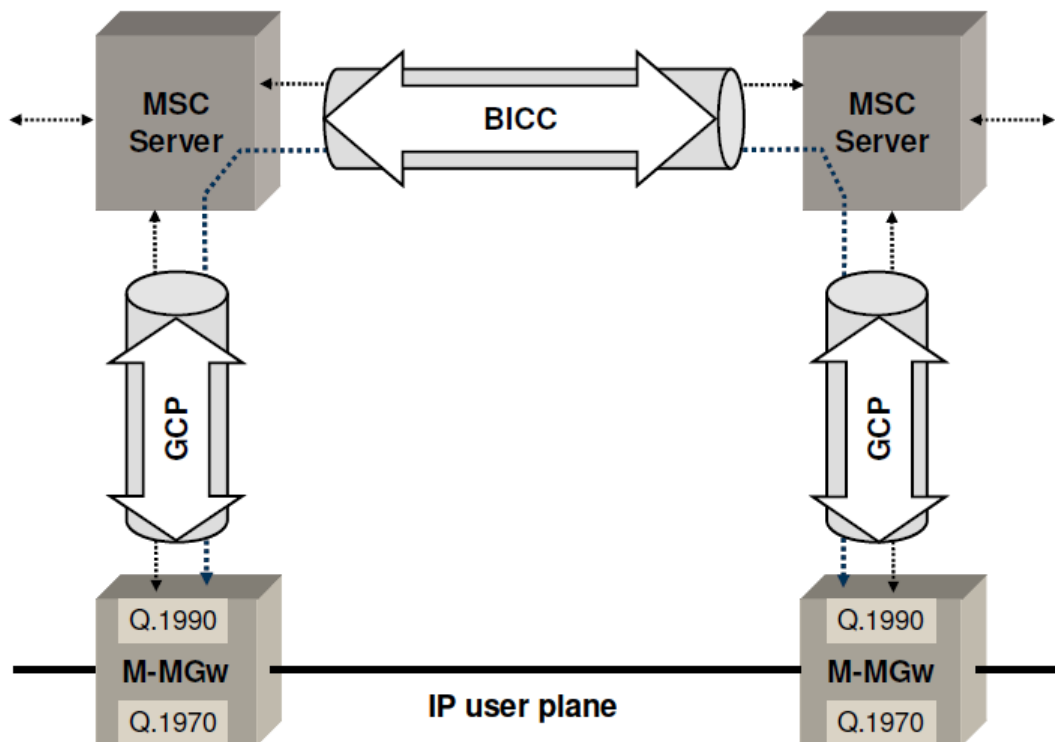


Figure 4.7 IPBCP Protocol (GSM mobile softswitch solution R3.1 introduction 2006:67).

4.3.8.1 IP Bearer Control Protocol (Q.1970)

The Q.1970 is an IP bearer control protocol. The IP bearer control protocol uses another protocol. The Session Description Protocol (SDP) is used for encoded the media stream information, such as port number and IP address. IP bearer is a bidirectional user plane that carrying media stream information between two Bearer Interworking Functions (BIWFs) across IP networks. Two types of BIWFs one is Initiating Bearer Interworking Function (I-BIWF) another one is Receiving Bearer Interworking Function (R-BIWF). In M-MGw initiating establishment of IP bearer is called I-BIWF. And in the M-MGw the receiving the establishment request of IP bearer is called R-BIWF. In IPBCP uses message to exchange information between BIWFs, there are four types of messages are as follows (ITU-T Q.1970 2001:1-4; Ericsson AB 2006:69).

- Request

The request message is sends to the BIWF that initiate the establishment of IP bearer it is known as an I-BIWF.

- Accept

The accept message is sends to the BIWF that receives the establishment of IP bearer is know as an R-BIWF.

- Confused

The confused message is sends to the BIWF of an IP bearer establishment request but it does not provide the receive request message.

- Reject

The reject message is sends to the BIWF of an IP bearer establishment request but when it rejects the request.

4.3.8.2 BICC Bearer Control Tunneling Protocol (Q.1990)

“The BICC bearer control tunneling protocol is a generic tunneling mechanism for the purpose of tunneling Bearer Control Protocols (BCP). The Bearer Control tunneling protocol (BCTP) transports tunneled Protocol Data Units of the supported bearer control protocols (BCP)”(Ericsson AB 2006:70). The Bearer control tunneling protocol used BICC Application Transport Mechanism (APM) and Call Bearer Control (CBC) protocols for tunneled. The bearer control information referred as BCTP PDUs and BCTP PDUs has two things one is BCTP indicator Field and IPBCP Session Description Protocol (SDP). In Figure 4.8 describes the Bearer Control Information structure (ITU-T Q.1990 2001:1; Ericsson AB 2006:70).

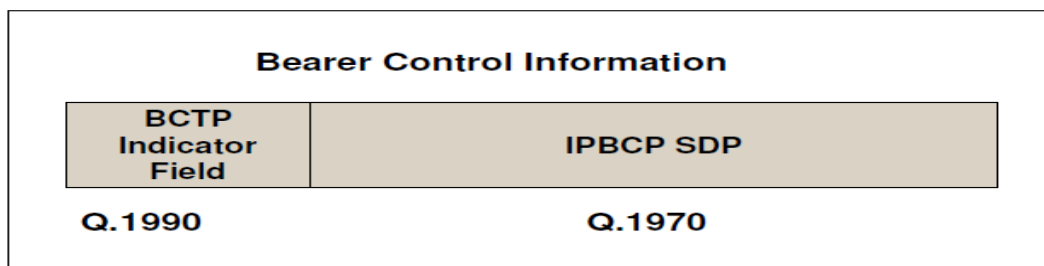


Figure 4.8 Bearer Control Information (GSM mobile softswitch solution R3.1 Introduction 2006:70).

4.4 Gateway Control Protocol (GCP/H.248)

H.248 protocol works as a master-slave protocol. The Media Gateway Controller (MGC) work as a master that control the Mobile Media Gateway (M-MGw) or Media Gateway (MG) work is known as slaves. M-MGw executes MGC instruction and answer can be done based on notifications. Figure 4.9 shows the structure of Gateway Control Protocol (Kopajtic & Lusa 2003:677; Ericsson AB 2006:58).

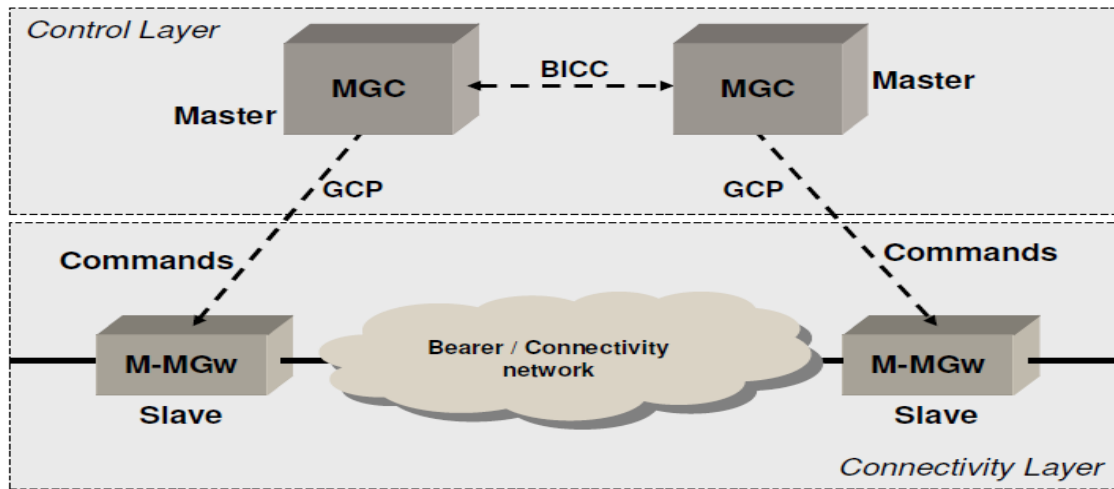


Figure 4.9 Gateway Control Protocol (GSM mobile softswitch solution R3.1 introduction 2006:58).

4.4.1 Protocol Connection Model (GCP/H.248)

The Protocol Connection model consist of three main components such as (Kopajtic & LuSa 2003:677; Ericsson AB 2006:59).

- Terminations
- Contexts
- Media Streams

4.4.1.1 Termination

A termination refers the source or destination of one or more streams where the media streams parameters and the bearer connection parameters are encapsulated. Figure 4.10 describe the connection model (Kopajtic & LuSa 2003:677; Ericsson AB 2006:59). H.248 or GCP has two types of terminations one is Physical termination another one is Ephemeral

termination. In the physical termination has the fixed capacity device such as termination 64 Kb/s timeslots and E1 or T1. If any users use a physical termination then no other user shares that resource (Ericsson AB 2006:60).

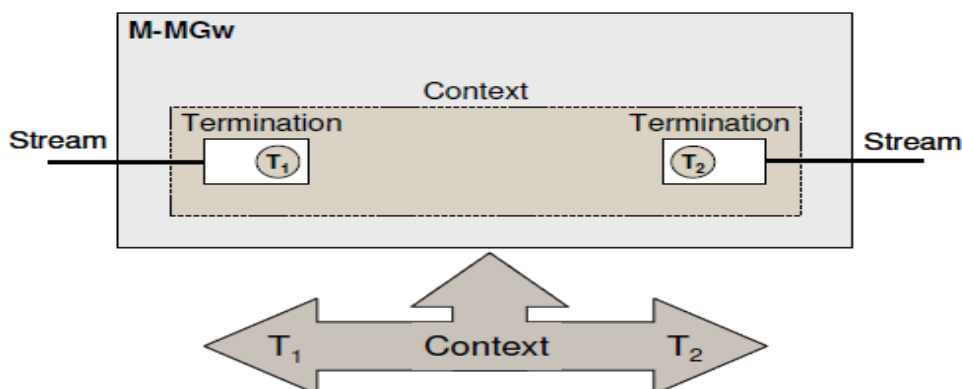


Figure 4.10 Connection Model (GSM mobile softswitch solution R3.1 introduction 2006:59).

In Ephemeral termination works as logical resources, users can share IP capacity from the M-MGw. When the IP connection established, the M-MGw assigns the capacity through physical interface and also returns a reference number. This IP termination is called Ephemeral termination. This termination can be done from the command of GCP. Figure 4.11 shows the Physical and Ephemeral Termination (Ericsson AB 2006:60).

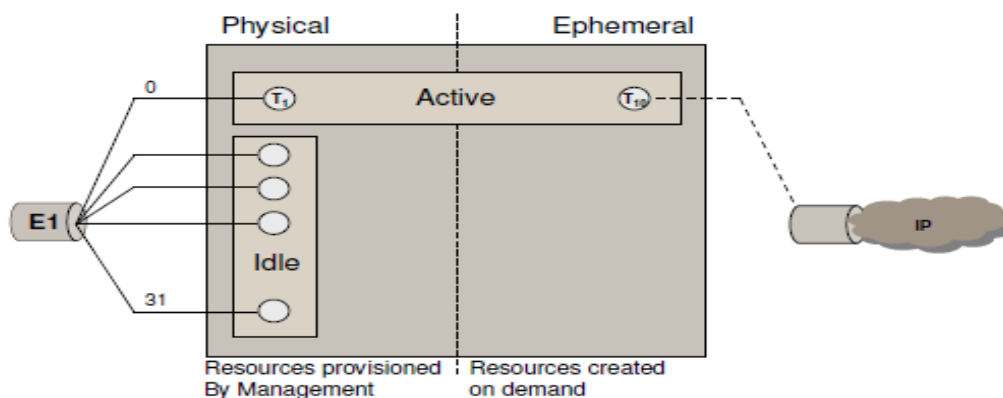


Figure 4.11 Physical and Ephemeral Termination (GSM mobile softswitch solution R3.1 introduction 2006:60).

4.4.1.2 Contexts

A context is a connection between one or more terminations. Every termination must have one context. In the physical termination the idle line represents a special context. It is called NULL context. After the command from MGC the M-MGw uses a specific physical termination and also set a NULL context. Figure 4.12 shows context and NULL context (Kopajtic & LuSa 2003:677; Wanjiun, Jen-Chun & Li 2005: 952; Ericsson AB 2006:59).

4.4.1.3 Media Stream

Information that contains in a single connection where the termination provides terminates or originates facilities is known as a media stream in the GCP (Ericsson AB 2006:61).

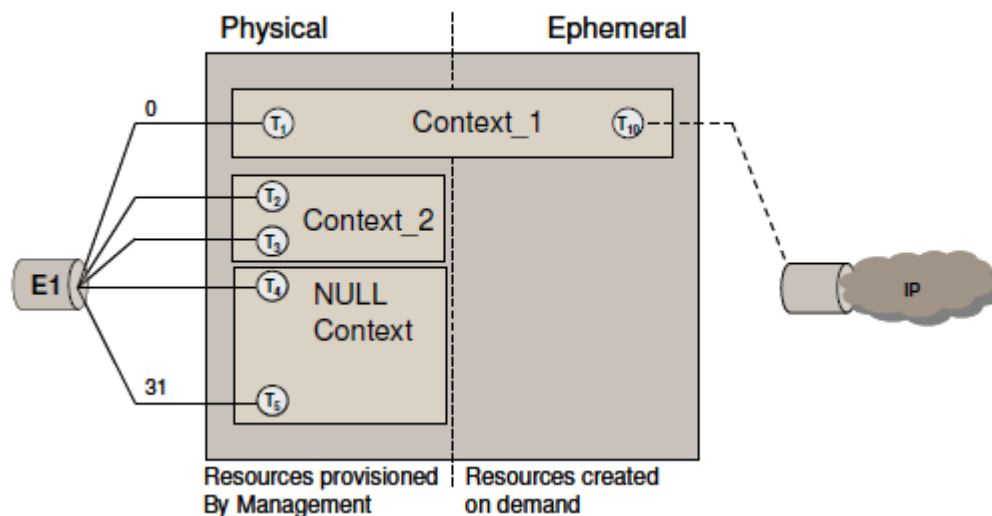


Figure 4.12 Context and NULL context (GSM mobile softswitch solution R3.1 introduction 2006:61).

4.4.2 Command in H.248 or GCP

There are eight different commands are using in H.248/GCP there are the given bellow

- Add
- Subtract
- Move
- Modify
- Audit value
- Service change
- Notify and
- Audit capabilities

The Add, Move and subtract commands are used for adding, removing and subtract termination corresponds on the context. Notify command means if it gets specific information then it notify by the M-MGw and MGC. Figure 4.13 shows the call processing in H.248/GCP connection creation and connection termination procedure (Wanjiun et al. 2005: 952; Ericsson AB 2006:62).

This is the simple example of point to point two party's calls where the MGC1 and the MGC2 are the MGCs. Those work as a master mode and that controls the slave MG1 and MG2 respectively. Being Notify to the MG1 that connection has created then the MGC1 sends the add command to the MG1 to create a calling termination. This can be done by "recvonly" operational mode because of the calling termination only receives the stream from the connected end. "Recvonly" is a unidirectional receive-only parameter. After that the MG1 sends the reply from calling termination to MGC1. This reply message also contains the modification of local descriptor which includes the IP address of MG1, RTP port number and session descriptor uses SDP. After that receiving the reply MGC1 it checks the routing table that contains the information where the outgoing gateway that the

called party is located. For example the outgoing gateway is MG2. Then it notifies to the MGC2 connection has created then the same procedure done the MGC2 where the MGC1 has done.

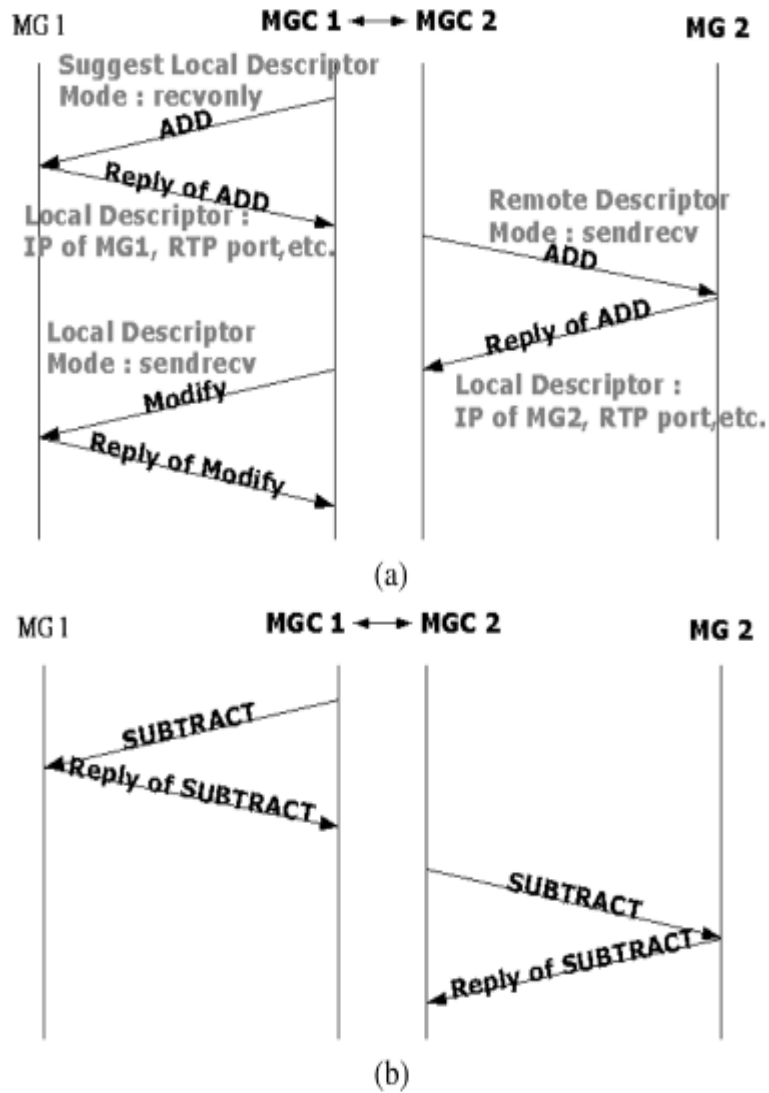


Figure 4.13 Call processing in G.248/GCP (a) Connection creation (b) Connection termination (Application-Layer Conference Trees for Multimedia Multipoint Conferences Using Megaco/H.248 2005:952).

The MGC2 sends the add command with the description that receives from the MGC1. This is done for create a called termination to the MG2. Then the called termination responds what the MGC2 descriptor then again the reply procedure. After that the MGC1 forwards the descriptor from called party to the calling party. Termination using modifies command changes the operation mode also as a “sendrecv”. Then the calling termination quickly replies for confirmation that the bi-directional connection established. In Figure 4.13 (b) describe the call termination procedure for that the MGCs send the subtract command to the MGs. These are done because clearing the calls and also free the resources that allocated for termination (Wanjiun et el. 2005: 952-953).

4.5 NBUP Protocol

The NbUp protocol is generally found in the user plane of circuit Switch (CS) core network it is work over Nb interface for example M-MGw to M-MGw. Generally it is initiated one of the M-MGw and tries to acknowledge to other M-MGw. In Figure 4.14 shows the NbUp protocol (Ericsson AB 2006:74).

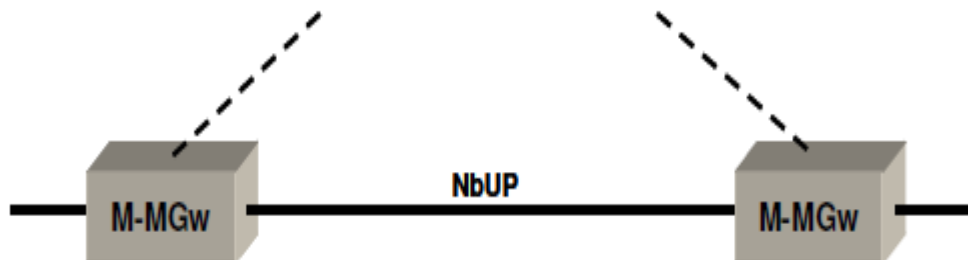


Figure 4.14 NbUp (GSM mobile softswitch solution R3.1 introduction 2006:74).

4.5.1 NbUp for Transport Layer

To interconnecting M-MGws IP network can be used for this NbUp can transport through IP bearer (Ericsson AB 2006:75).

4.5.1.1 Transport NbUp over IP

For transport NbUp over IP network, RTP over UDP over IP (IPV4 or IPV6) shall be used. Figure 4.15 shows the protocol stack for transport network user plane on the Nb interface. For established IP bearer IPBCP shall be used then tunneled the IPBCP messages through GCP and BICC (3GPP TS 29.414V7.0.0 2005:11; Ericsson AB 2006:75-76).

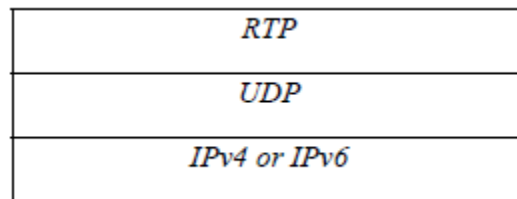


Figure 4.15 IP Protocol stack for the transport network user plane (3GPP TS 29.414 V7.0.0 2005)

UDP port number used in every M-MGw for established the RTP bearer connection that can transport in a singled NbUp connection. The port number is assign for the RTP protocol. In the M-MGw same port numbers used to defined send and receive RTP PDUs (3GPP TS 29.414V7.0.0 2005:11; Ericsson AB 2006:75-76).

4.6 Call Setup example in an IP Network

Figure 4.16 and 4.17 shows simplified call flow in Mobile softswitch solution that shows the message flow for a forward bearer establishment procedure over an IP network (Ericsson AB 2006:77-80).

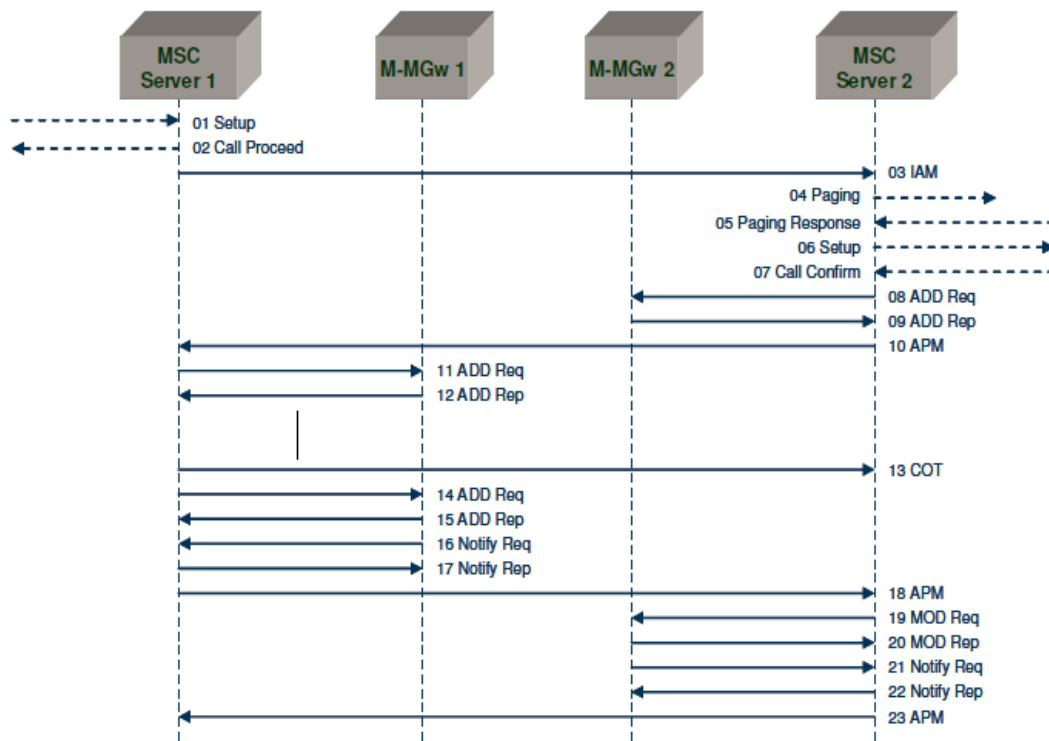


Figure 4.16 Simplified Call flow in MSS (GSM mobile softswitch solution R3.1 introduction 2006:77).

- 1) First the Mobile Station (MS) sends the setup message to the MSC server 1.
- 2) The MSC server 1 response with sending call precede message.
- 3) The MSC server 1 sends Initial Address Message (IAM) to the MSC server 2 that indicated forward bearer establishment and tunneling.

- 4) Then the MSC server to start paging to the Mobile Station (MS).
- 5) Then the Paging response request sends back.
- 6) MSC server then sends a setup message.
- 7) The Mobile Station (MS) sent a call confirm message to the MSC server 2.
- 8) The MSC server 2 then sends add request to the M-MGw 2 to the core side for preparing the bearer setup message.
- 9) Then the M-MGw 2 answers the added reply message.
- 10) The MSC server 2 sent Application Transport message (APM) to the MSC server 1 inside the APM message contains the identity (BCUID) of the selected M-MGw2.
- 11) MSC server 1 sends add request to the M-MGw 1 for bearer preparation.
- 12) Then M-MGw 1 sends add reply message to the MSC server 1. Signaling to the BSC left out.
- 13) Then the MSC server 1 sends the continuity (COT) message to the MSC server 2.
- 14) MSC server 1 then sends the Add request message to the core network for bearer establishment.
- 15) M-MGw 1 then sends the confirmation with Add reply message to MSC server 1.
- 16) The M-MGw 1 sends the notify request message to the MSC server 1 and also M-MGw 1 tunnel the message IPBCP.
- 17) Then the MSC server 1 sends the confirmation message with Notify reply to the M-MGw 1.
- 18) The MSC server 1 used the Application Transport (APM) message to tunnel the request message to the MSC server 2.
- 19) Then the request message tunneled down sent to the M-MGw 2 using a Modify request message.
- 20) The confirmation of the message from M-MGw 2 sends with Modify reply message to MSC server 2.
- 21) The M-MGw 2 sends Notify request message that tunnels an accept message IPBCP.
- 22) Notify reply sends from MSC server 2 to M-MGw 2.

23) Then the Application Transport (APM) message sends from MSC server 2 to MSC server 1.

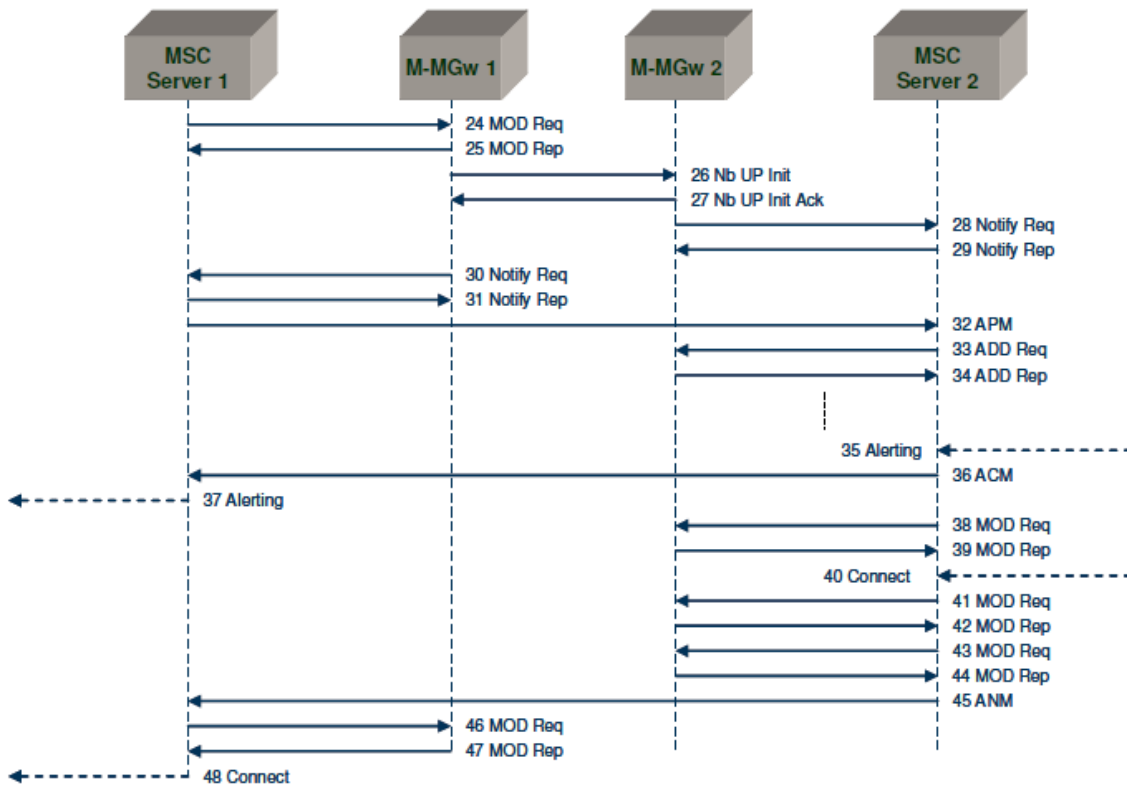


Figure 4.17 Simplified Call flow in MSS (GSM mobile softswitch solution R3.1 introduction 2006:79).

24) The MSC server 1 sends the Modify request to the M-MGw 1 to tunnel the IPBCP accepted message.

25) The M-MGw 1 sends the confirmation with Modify replay message to the MSC server 1.

26) Now the communication between the user part M-MGw1 and M-MGw 2 for this the M-MGw 1 sends the NbUp init message to the M-MGw 2.

- 27) The M-MGw 2 then confirms the message by sending NbUp init acknowledgement message to the M-MGw 1.
- 28) The M-MGw 2 sends Notify request to the MSC server 2 which means that the user part establishment has done.
- 29) MSC server 2 then sends the confirmation by sending Notify reply message.
- 30) The M-MGw 1 sends Notify request to the MSC server 1 which means that the user part establishment has done.
- 31) MSC server 1 then sends the confirmation by sending Notify reply message.
- 32) The MSC server 1 then sends APM message to the MSC server 2.
- 33) The MSC server sends Add request message to the M-MGw 2 for reserve a termination.
- 34) The M-MGw 2 sends the confirmation by sending Add reply message to MSC server 2, again the BSC specific signaling is left out.
- 35) Alerting is received from Mobile Station (MS).
- 36) The MSC server 2 sends the Address Complete (ACM) message to the MSC server 1.
- 37) The MSC server 1 sends alerting message to the Mobile Station (MS).
- 38) MSC server 2 then sending Modify request to the M-MGw 2 for sending the tone.
- 39) Then the M-MGw confirm by sending the Modify reply message.
- 40) The MS indicates the connection to MSC server 2.
- 41) The MSC server 2 sends Modify request to the M-MGw 2 which indicates stop the tone.
- 42) Then the M-MGw2 confirmations by sending the Modify reply message.
- 43) The MSC server 2 sends Modify request to the M-MGw 2 for inside connection of M-MGw 2.
- 44) Then M-MGw 2 confirms by sending the Modify reply message.
- 45) After that through the connection the MSC server 2 sends the Answer Message (ANM) to the MSC server 1.

- 46) The MSC server 1 sends Modify request to the M-MGw 1 for the inside connection M-MGw 1.
- 47) Then M-MGw 1 confirms by sending the Modify reply message.
- 48) Finally, the MSC server 1 sends the Connect message to the Mobile Station (MS).
So the call is established now.

5. PERFORMANCE AND ARCHITECTURE OVERVIEW OF SOFTSWITCH NETWORKING

Depending on different management of user's information and the signaling and media connectivity, softswitch networking architecture can be partitioned in the following ways (Phromsuphorn, Koseeyaporn & Wardkein 2008:534; Xu et al. 2005:54).

- Single Layer Architecture (SLA)
- Subscriber Management and Signaling Routing Layered Architecture (SMSRLA)
- Fully Layered Architecture (FLA)

5.1 Single Layer Architecture (SLA)

The SLA is the simplest network architecture. In this architecture the part of signaling, the call setup will take only one hop. To perform the task of SLA, all softswitch should maintain signaling routing information. When a new subscriber connects to the network then the database should be updated. For this process the data always passes through IP trunk and when the number of softswitch increases, the IP trunk also increases exponentially. Figure 5.1 shows the SLA softswitch network. The softswitch working method divides into two parts. First one is the Call Control (CC) part, where the call control operations are structured and ensure that the call control messages are properly passed. Another one is the Subscriber Management & Routing (SMR) that take care the message management such as the authorization, authentication and signaling routing. In Figure 5.2 shows the softswitch function diagram and Figure 5.3 shows softswitch queuing model (Phromsuphorn et al. 2008:534; Xu et al. 2005:54-55).

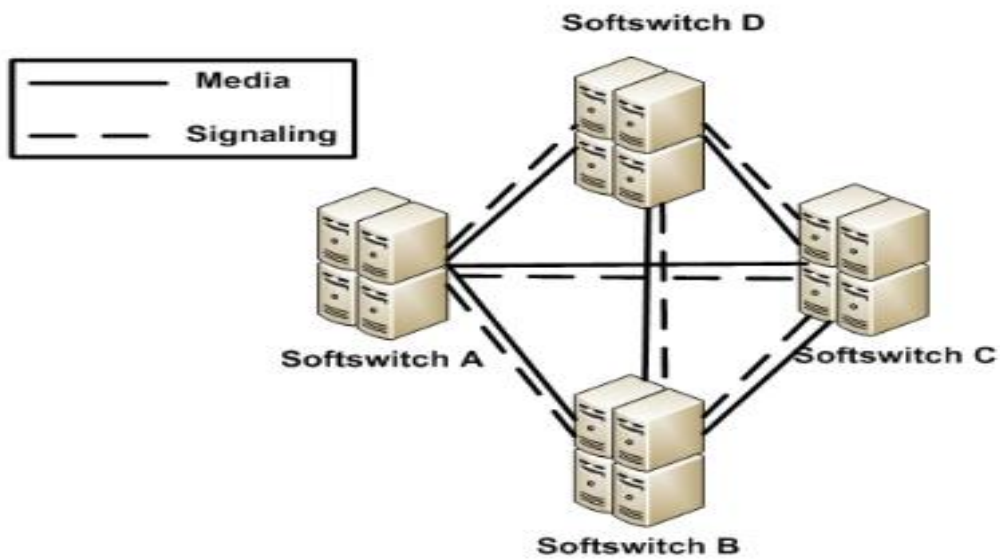


Figure 5.1 SLA of Softswitch network (Performance Analysis of Soft Switch Network Based on Jackson Network Theory 2008:534).

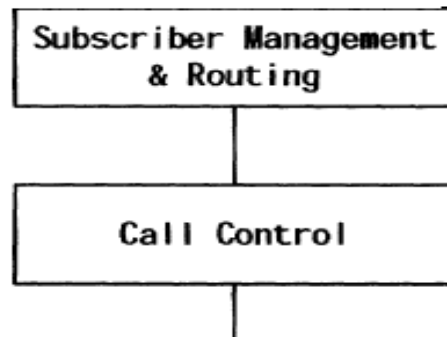


Figure 5.2 Softswitch Functional block (Architecture and Performance of Softswitch Networking 2005:55).

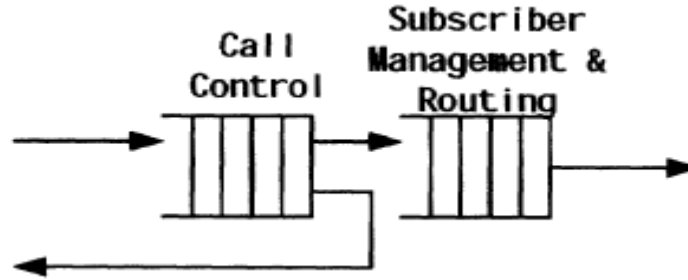


Figure 5.3 Softswitch Queuing network model (Architecture and Performance of Softswitch Networking 2005:55).

In Figure 5.3 Call control queue and Subscriber Management & routing queue are assumed as M/M/1 queuing model. The output of call control process is the input of SMR. To find the call setup delay in the softswitch, need to know the average service time of the CC is assumed to be C and also same as SMR assumed it is τ . τ will increase when the subscriber management are increasing by a softswitch. So that it can be given that $\tau=f(m)$ where m represents the subscriber number. Now the following are the parameters that we need to know to find out the call setup delay in Single Layer Architecture (SLA) (Phromsuphorn et al. 2008:534; Xu et al. 2005:54-55).

- The arriving call setup requests will increase linearly in the softswitch. Accordingly, increasing the number of users are connected directly to the softswitch. Assume λ is the call setup request then $\lambda=\alpha m$, where m is the number of subscribers and α can be described as follow based on traffic theories of Telecommunication

$$\alpha = \frac{\text{Traffic single Subscriber}}{\text{Time length Average call}}.$$

Where the traffic single subscriber is the traffic of single subscriber and the Time length average call refers the average length of time for a call.

- r is defined as a ratio where the total number of outgoing calls from the total number of calls. r can be found from the table arriving rate of each queue.
- The number of subscribers that are directly connected to softswitch 1 and softswitch 2 is m .

- The transfer delay of signaling message over IP backbone is d .
- Finally, the entire number of subscribers in the softswitch network is M .

Depending on those above assumption the SLA model in Figure 5.4 that represents the queuing model shows in Figure 5.5.

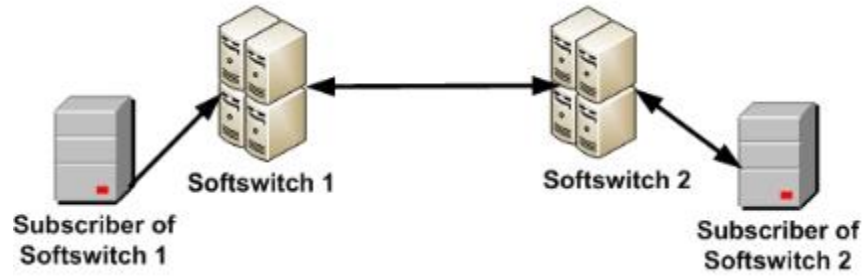


Figure 5.4 SLA model (Architecture and Performance of Softswitch Networking 2005:55).

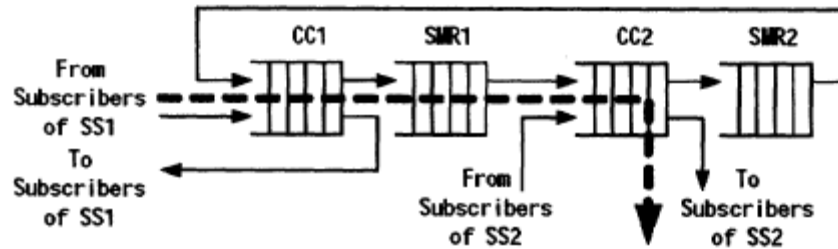


Figure 5.5 Softswitch Queuing network model for SLA (Architecture and Performance of Softswitch Networking 2005:55).

In Figure 5.5 the dashed line shows the path of call setup message to pass the softswitch network. The call setup delay T represents the following equation (Xu et al. 2005:54-55; phromsuphorn et al. 2008:534).

$$T = T_{cc1} + T_{SMR1} + T_{cc2} + d \quad (1)$$

In the equation (1), T_{CC1} , T_{SMR1} and T_{CC2} are represented the delay of softswitch queuing model CC1, SMR1 and CC2. The arriving rate and the service rate for each queue in Figure 5.5 can be found in the Table 1 and Table 2 (Bertsekas & Gallager 1992:129).

λ_{CC1}	λ_{SMR1}	λ_{CC2}	λ_{SMR2}
$(1+r)\alpha m$ where $\lambda = \alpha m$	$r\alpha m$ where $\lambda = \alpha m$	$(1+r)\alpha m$	$r\alpha m$

Table 1. Arriving rate of each queue system.

μ_{CC1}	μ_{SMR1}	μ_{CC2}	μ_{SMR2}
$\frac{1}{C}$	$\frac{1}{f(M)}$	$\frac{1}{C}$	$\frac{1}{f(M)}$

Table 2. Service rate of every queue system.

Now to find the delay of CC1 (T_{CC1}) as an M/M/1 queuing model the offered traffic is,

$$\rho = \frac{\lambda}{\mu} \quad (2)$$

Queuing time, from Little's law,

$$\tilde{N} = \lambda T \quad (3)$$

Then we can get,

$$T = \frac{\tilde{N}}{\lambda} \quad (4)$$

Where \tilde{N} is the average number of calls in the system that can be represents as,

$$\tilde{N} = \frac{\rho}{(1-\rho)} \quad (5)$$

$$T = \frac{\tilde{N}}{\lambda} = \frac{\rho}{\lambda(1-\rho)} \quad (6)$$

Using $\rho = \frac{\lambda}{\mu}$ then queuing time,

$$T = \frac{1}{(\mu - \lambda)} \quad (7)$$

For queuing time CC1 can be found in the given bellow equation (Phromsuphorn et al. 2008:534; Xu et al. 2005:54-55).

$$T_{cc1} = \frac{1}{(\mu_{cc1} - \lambda_{cc1})} = \frac{C}{1 - C\alpha m(r+1)} \quad (8)$$

Same way uses to find out the delay equation for T_{SMR1} and T_{CC2}

$$T_{SMR1} = \frac{1}{(\mu_{SMR1} - \lambda_{SMR1})} = \frac{f(M)}{1 - r\alpha m f(M)} \quad (9)$$

$$T_{cc2} = \frac{1}{(\mu_{cc2} - \lambda_{cc2})} = \frac{C}{1 - C\alpha m(r+1)} \quad (10)$$

Now substituting (8), (9) and (10) equation into (1) the call setup delay for SLA can be represented as follow,

$$T = \frac{2C}{1 - C\alpha m(r+1)} + \frac{f(M)}{1 - C\alpha m(r+1)} + d \quad (11)$$

5.2 Subscriber Management and Signaling Routing Layered Architecture (SMSRLA)

Routing Server (RS) is used in SMSRLA, which handles the users and signaling routing information in softswitch network. Moreover it provides the routing facility for service subscriber. RS does not manage information in details as that of softswitch. It only keeps track certain subscribers and the softswitch can access those subscribers directly. In RS, E.164 prefix and IP subnet address used to manage the subscriber and signaling information. E.164 prefix refers the global numbering plan where the device is connected to the telephone networks. This assigned a unique numerical address. The International Telecommunication Union (ITU) is administrated the global numbering plan. RS can manage large quantity of subscribers then that of Softswitch. Figure 5.6 shows the SMSRLA (Xu et al. 2005:56 ; Phromsuphorn et al. 2008:536; Cisco online).

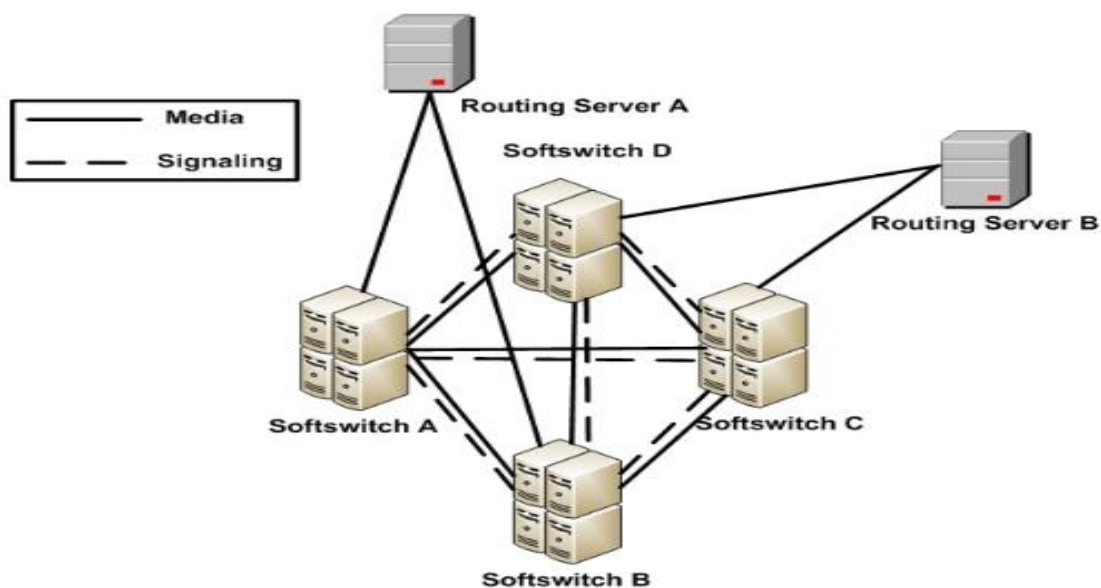


Figure 5.6 Subscriber Management and Signaling Routing Layered Architecture (Architecture and Performance of Softswitch Networking 2005:56).

Signaling routing can be done in two ways in SMSRLA one is signaling routing proxy other one is signaling routing redirection. In signaling routing proxy, the RS work as a proxy and forwarding signaling message between two softswitches. In signaling routing redirection the RS helps the softswitch to execute signaling routing. The call setup delay of SMSRLA can be shown in Figure 5.7. The RS is very simple so that the M/M/1 queuing model used. Figure 5.8 shows the queuing model for SMSRLA the dashed line is known as call setup direction of the softswitch network (Xu et al 2005:56 ; Phromsuphorn et al 2008:536; Cisco online).

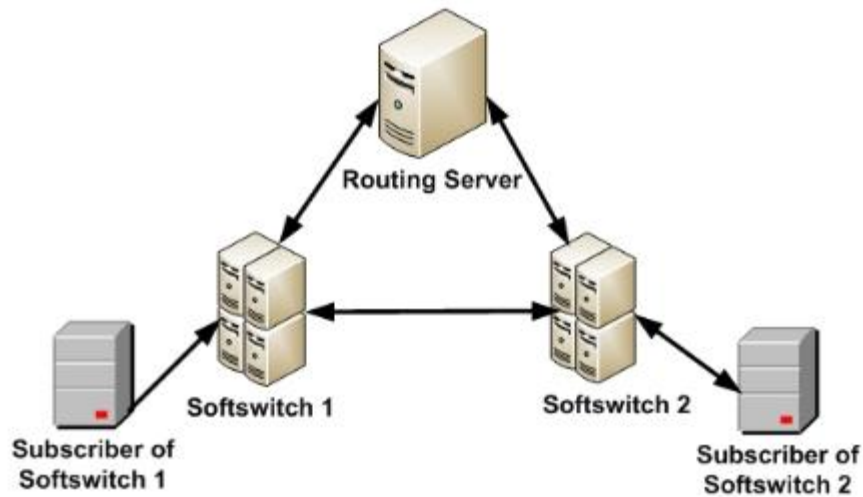


Figure 5.7 SMSRLA model (Architecture and Performance of Softswitch Networking 2005:56).

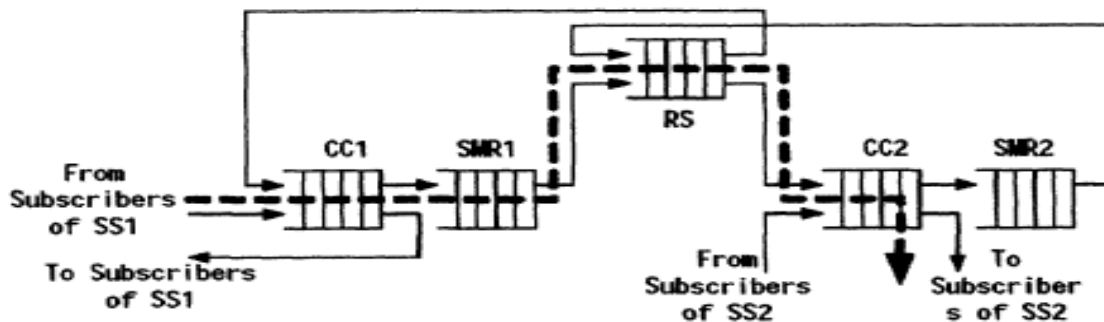


Figure 5.8 Queuing network model for SMSRLA (Architecture and Performance of Softswitch Networking 2005:56).

Finally, the call setup delay can represent the following equation for SMSRLA,

$$T = T_{cc1} + T_{SMR1} + TRS + T_{cc2} + 3d \quad (12)$$

The arriving rate and the service rate for every queue for Figure 5.8 can be found in the Table 3 and Table 4.

λ_{CC1}	λ_{SMR1}	λ_{RS}	λ_{CC2}	λ_{SMR2}
$(1+r)\alpha m$ where $\lambda = \alpha m$	$r\alpha m$	$2r\alpha m$	$(1+r)\alpha m$	$r\alpha m$

Table 3. Arriving rate of every queue system

μ_{CC1}	μ_{SMR1}	μ_{RS}	μ_{CC2}	μ_{SMR2}
$\frac{1}{C}$	$\frac{1}{f(m)}$	$\frac{1}{f(M, m)}$	$\frac{1}{C}$	$\frac{1}{f(m)}$

Table 4. Service rate of every queue system

The RS can handle more subscribers then the softswitch. For an example, one softswitch can handle 10000 subscribers but the RS could connect more then one softswitch.

The RS average service time is $f(M, m)$ so, the service rate is,

$$\frac{1}{f(M, m)} \quad (13)$$

Where M is the total number of subscribers that is connected to the softswitch network and m is the number of subscriber that connected directly one softswitch. Now we can find call

setup delay for SMSRLA in the same way as SLA (Phromsuphorn et al. 2008:536; Xu et al. 2005:56;Cisco online).

$$T_{cc1} = \frac{1}{(\mu_{cc1} - \lambda_{cc1})} = \frac{C}{1 - C\alpha m(r+1)} \quad (14)$$

Same way to find out the delay equation for T_{SMR1} , T_{RS} and T_{CC2}

$$T_{SMR1} = \frac{1}{(\mu_{SMR1} - \lambda_{SMR1})} = \frac{f(m)}{1 - r\alpha mf(m)} \quad (15)$$

$$T_{RS} = \frac{1}{(\mu_{RS} - \lambda_{RS})} = \frac{f(M, m)}{1 - 2r\alpha mf(M, m)} \quad (16)$$

$$T_{cc2} = \frac{1}{(\mu_{cc2} - \lambda_{cc2})} = \frac{C}{1 - C\alpha m(r+1)} \quad (17)$$

Now substituting (14), (15), (16) and (17) equation into (12) the call setup delay for SMSRLA can be represents as follow,

$$T = \frac{2C}{1 - C\alpha m(r+1)} + \frac{f(m)}{1 - r\alpha mf(m)} + \frac{f(M, m)}{1 - 2r\alpha mf(M, m)} + 3d \quad (18)$$

5.3 Fully Layered Architecture (FLA)

In Fully layered architecture two level of softswitch are used one is high level softswitch another one is low level softswitch. In high level softswitch manages to connect all the subscribers to their secondary softswitch. Whereas the low level softswitch manages to the subscriber directly connected to it. Figure 5.9 shows that signaling and media connection between two low level softswitch done via with the higher level softswitch. The

performance can examine from the block diagram 5.10 (Phromsuphorn et al. 2008:536; Xu et al. 2005:57).

Figure 5.10 the softswitch 0 refers to the higher level of softswitch which can perform call processing but not access the subscribers directly. Now the queuing part can proceed with the help from FLA block diagram. Figure 5.11 shows the queuing model in FLA (Phromsuphorn et al. 2008:536; Xu et al. 2005:57).

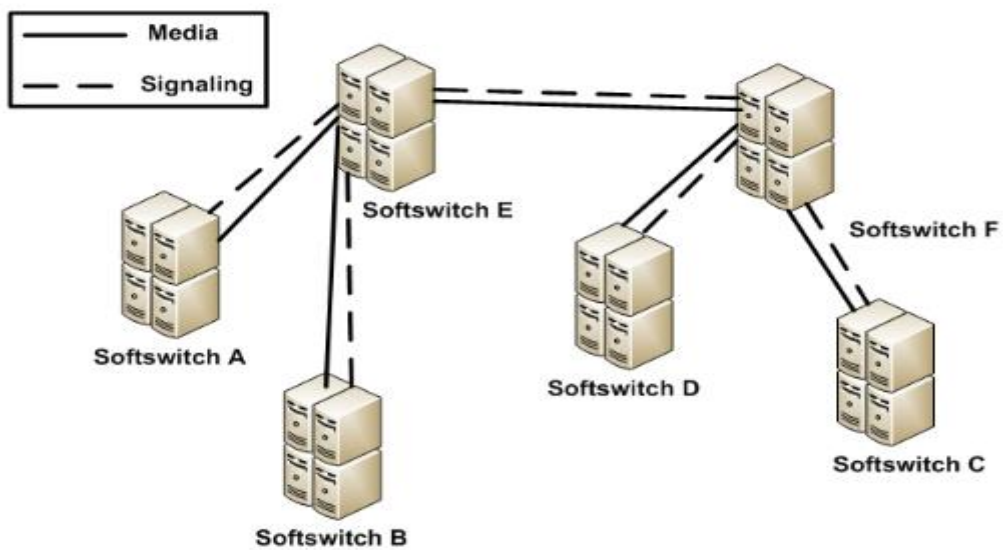


Figure 5.9 Fully Layered Architecture model (Performance Analysis of Soft Switch Network Based on Jackson Network Theory 2008:536).

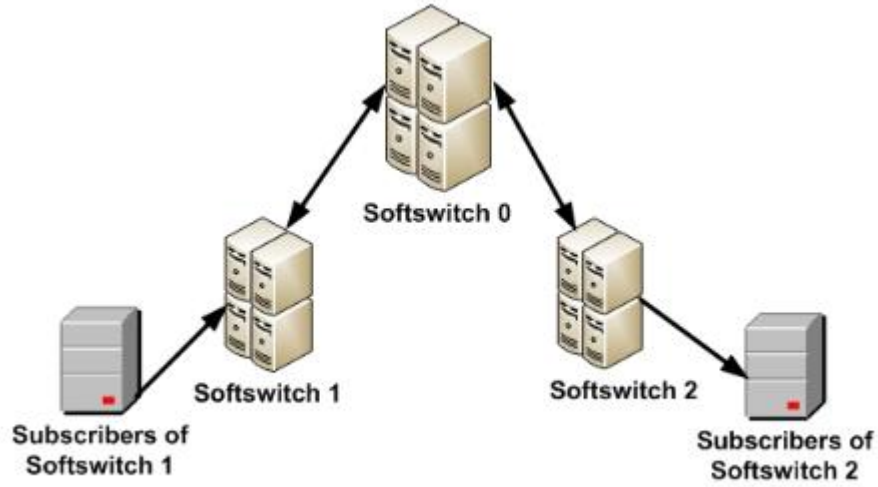


Figure 5.10 FLA block diagram (Performance Analysis of Soft Switch Network Based on Jackson Network Theory 2008:536).

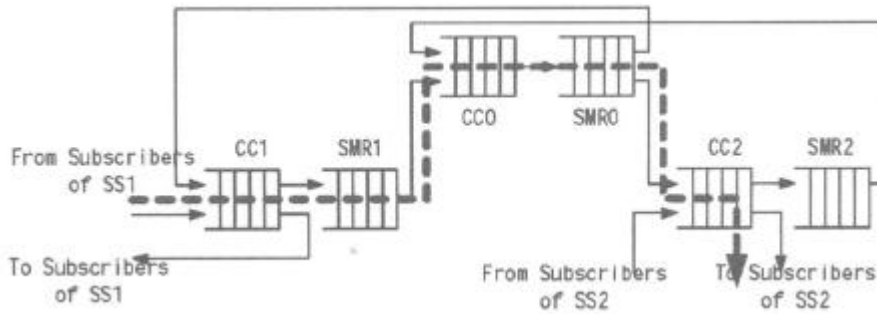


Figure 5.11 FLA queuing network model (Architecture and Performance of Softswitch Networking 2005:57).

Finally, the call setup delay can represent the following equation for FLA,

$$T = T_{cc1} + T_{SMR1} + T_{cc0} + T_{SMR0} + T_{cc2} + 2d \quad (19)$$

The arriving rate and the service rate for every queue for Figure 5.11 can be found in the Table 5 and Table 6.

λ_{CC1}	λ_{SMR1}	λ_{CC0}	λ_{SMR0}	λ_{CC2}	λ_{SMR2}
$(1+r)\alpha m$ where $\lambda = \alpha m$	$r\alpha m$	$2r\alpha m$	$2r\alpha m$	$(1+r)\alpha m$	$r\alpha m$

Table 5. Arriving rate of every queue system

μ_{CC1}	μ_{SMR1}	μ_{CC0}	μ_{SMR0}	μ_{CC2}	μ_{SMR2}
$\frac{1}{C}$	$\frac{1}{f(m)}$	$\frac{1}{C}$	$\frac{1}{f(M,m)}$	$\frac{1}{C}$	$\frac{1}{f(m)}$

Table 6. Service rate of every queue system

Now we can find call setup delay for FLA in the same way as before as in SLA and SMSRLA.

$$T_{cc1} = \frac{1}{(\mu_{cc1} - \lambda_{cc1})} = \frac{C}{1 - C\alpha m(r+1)} \quad (20)$$

Same way use to find out the delay equation for T_{SMR1} , T_{CC0} , T_{SMR0} and T_{CC2}

$$T_{SMR1} = \frac{1}{(\mu_{SMR1} - \lambda_{SMR1})} = \frac{f(m)}{1 - r\alpha m f(m)} \quad (21)$$

$$T_{cc0} = \frac{1}{(\mu_{cc0} - \lambda_{cc0})} = \frac{C}{1 - 2C r\alpha m} \quad (22)$$

$$T_{SMR0} = \frac{1}{(\mu_{SMR0} - \lambda_{SMR0})} = \frac{f(M,m)}{1 - 2r\alpha m f(M,m)} \quad (23)$$

$$T_{cc2} = \frac{1}{(\mu_{cc2} - \lambda_{cc2})} = \frac{C}{1 - C\alpha m(r+1)} \quad (24)$$

Now substitution (20), (21), (22), (23) and (24) equation into (19) the call setup delay for FLA can be represented as follow,

$$T = \frac{2C}{1 - C\alpha m(r+1)} + \frac{f(m)}{1 - r\alpha m f(m)} + \frac{f(M, m)}{1 - 2r\alpha m f(M, m)} + \frac{C}{1 - 2C\alpha m} + 2d \quad (25)$$

5.4 Simulation Analysis

In this part, the performance of different types of network shows the simulation by Matlab. For doing this the following parameter are applied (Xu et al. 2005:54-58; Best practices guide for developing spectral link 8020/8030 wireless telephones 2009).

- One softswitch could connect directly 10000 subscribers so m value is 10000.
- The outgoing calls ratio is 20 percents so r is 0.2.
- The average service time for CC part is 0.03s so C is 0.03s
- The average time to find out the subscriber information from database should increases linearly by the number of increasing subscriber. So SMR average time defined as $\tau = f(m) = am$ where a is 4.3×10^{-8} .
- To found out the α Traffic single subscriber is 0.10 Erlang and the Time length average call is 100s. One Erlang is equivalent to the traffic generated by a single telephone call in continuous use. A typical office telephone user will generate 0.10 to 0.15 Erlangs of normal work hours. 0.10 Erlang is generally used in telecommunication engineering. It is a common used engineering parameter so,

$$\alpha = \frac{\text{Traffic single Subscriber}}{\text{Time length Average call}} = \frac{0.10}{100} = 0.001$$

- The transfer delay of signaling message through IP backbone d is 0.025s.

- The RS and high level softswitch using E.164 prefix and to find out the subscriber information using the same algorithm to the subscriber database in the softswitch

$$\text{then, } f(M, m) = \frac{aM}{m}$$

Now substituting all those values into the (11), (18) and (25), which shows the relation between call setup delay and number of subscribers. In Figure 5.12 shows the relation between call setup delay and number of subscribers for SLA. The numbers of subscribers are ranging from 10^4 to 10^7 . For compare the performance between SMSRLA and FLA. In Figure 5.12, 5.13 and 5.14 shows when the subscribers ranging is less than 7×10^5 the best Solutions to select SLA. When the subscribers more than 9×10^5 the best solutions to select the SMSRLA and FLA (Phromsuphorn et al. 2008:536; Xu et al. 2005:57).

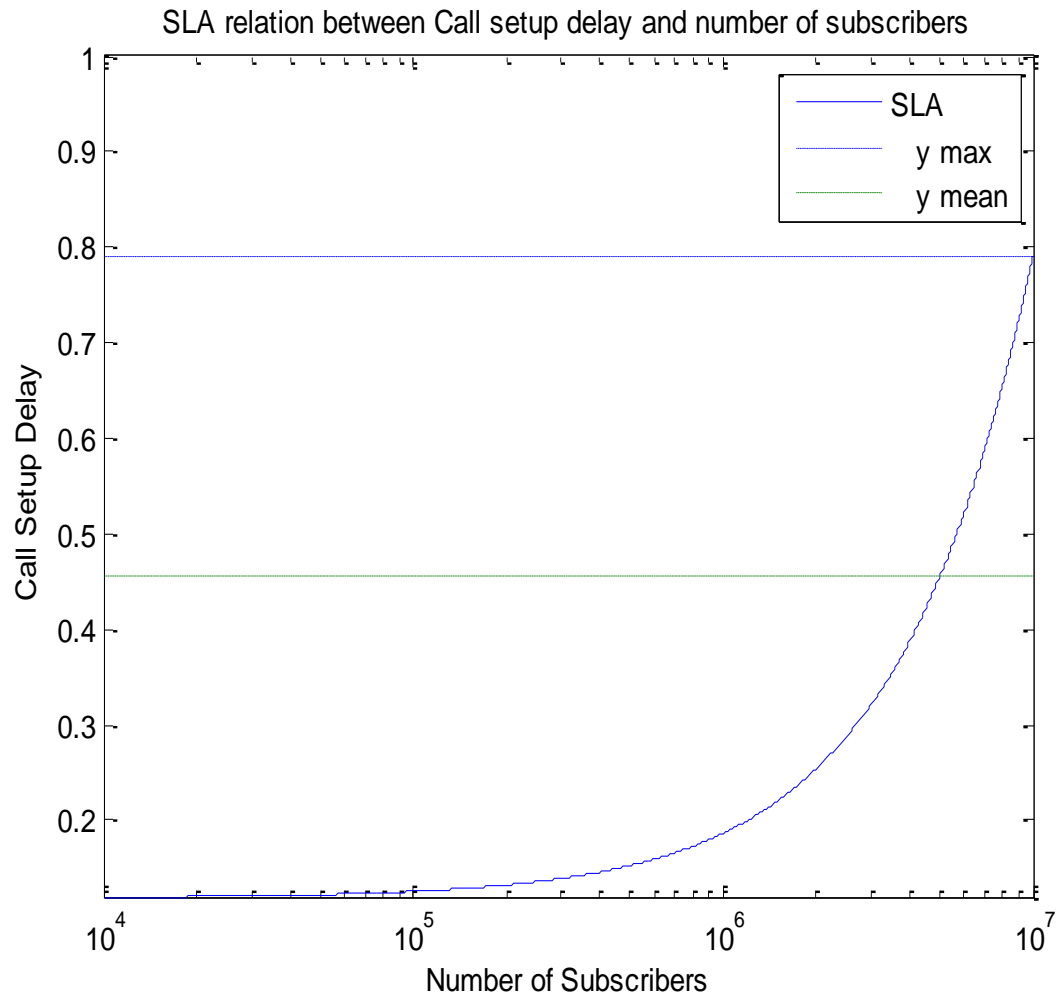


Figure 5.12 Relation between call setup delay and number of subscribers for SLA.

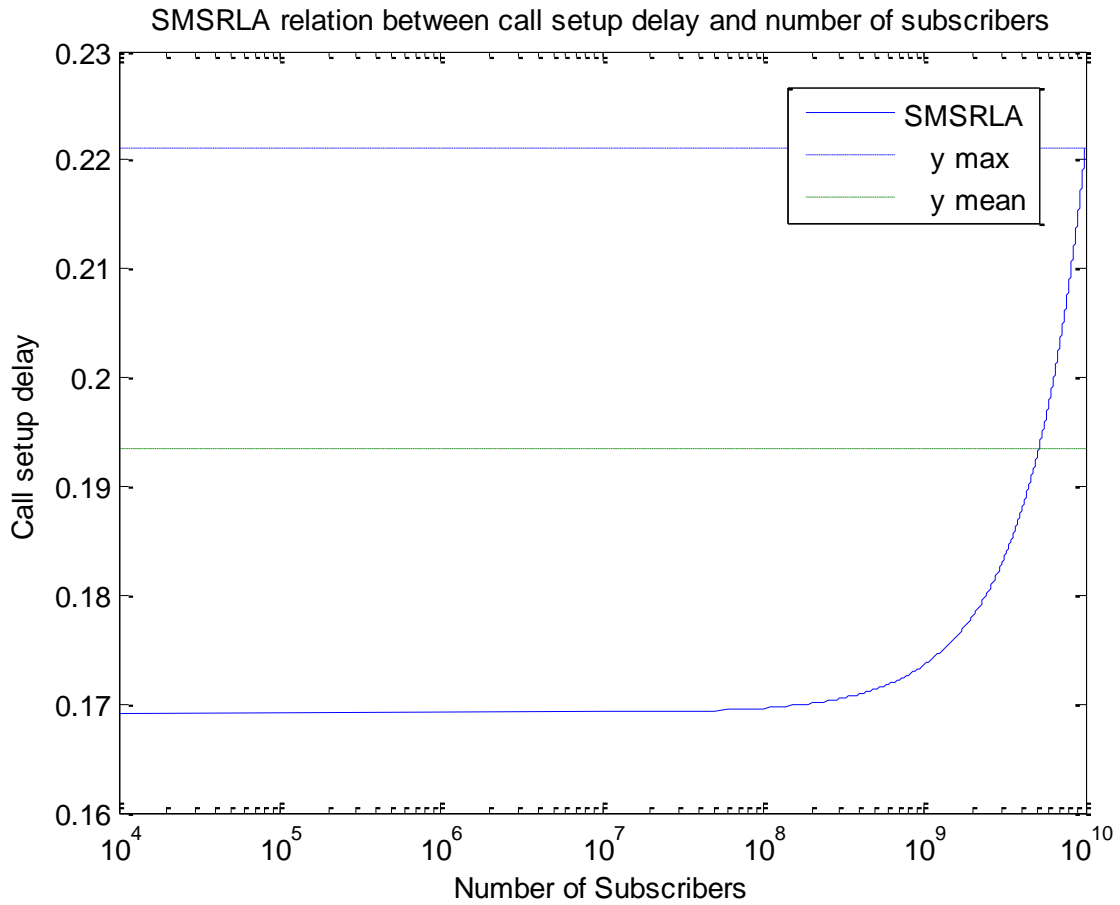


Figure 5.13 Relation between call setup delay and number of subscribers for SMSRLA.

Finally, to make decisions when the subscribers ranging more than 9×10^5 which network architecture should provides the best solutions comparing to the SMSRLA and FLA. When the transferring delay of signaling messages over IP backbone condition is really good as 0.025s than the SMSRLA is better than FLA which can shows in Figure 5.14.

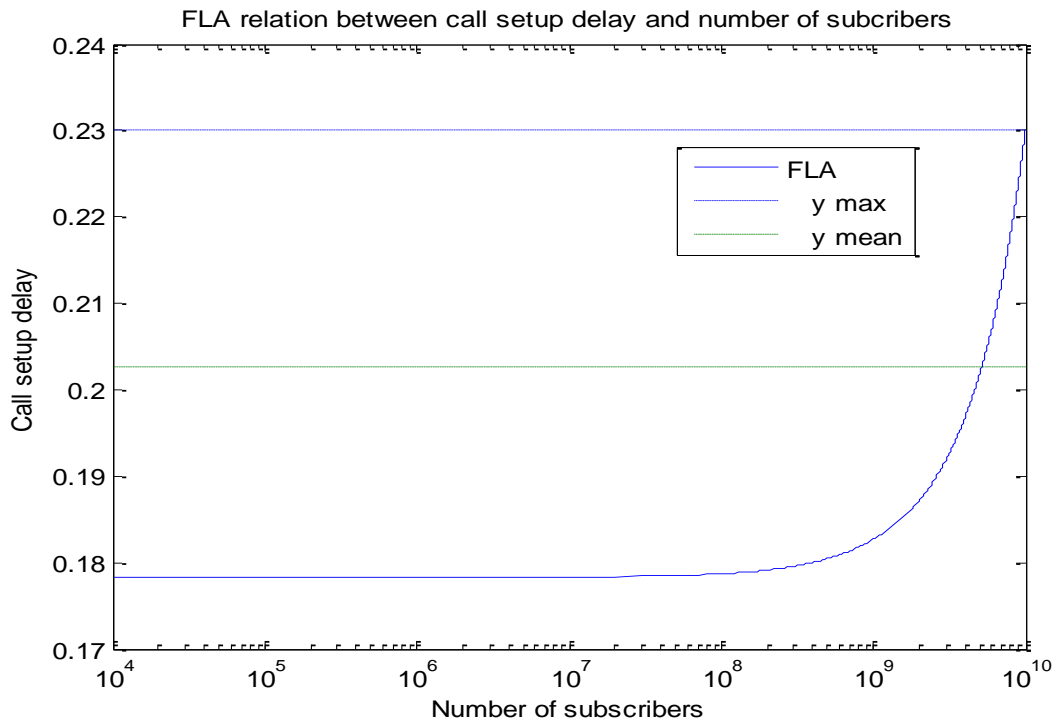


Figure 5.14 Relation between call setup delay and number of subscribers for FLA.

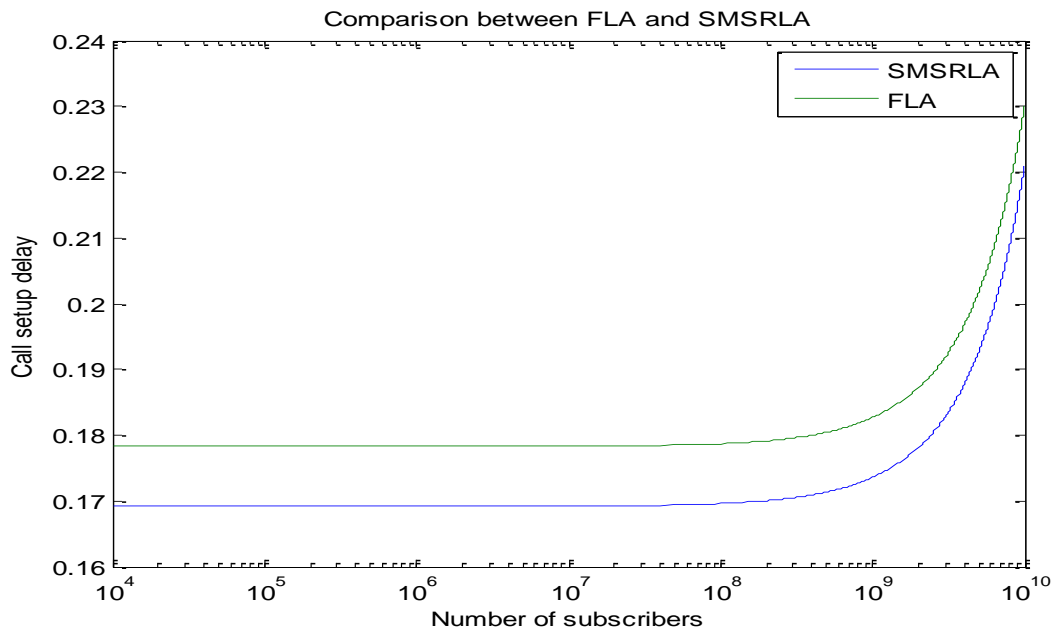


Figure 5.14 Comparison between FLA and SMSRLA performance.

Moreover, when considering the transferring signaling message to the IP backbone not good enough. For example delay is 0.045s than the performance of FLA is better than SMSRLA which can shows in Figure 5.15.

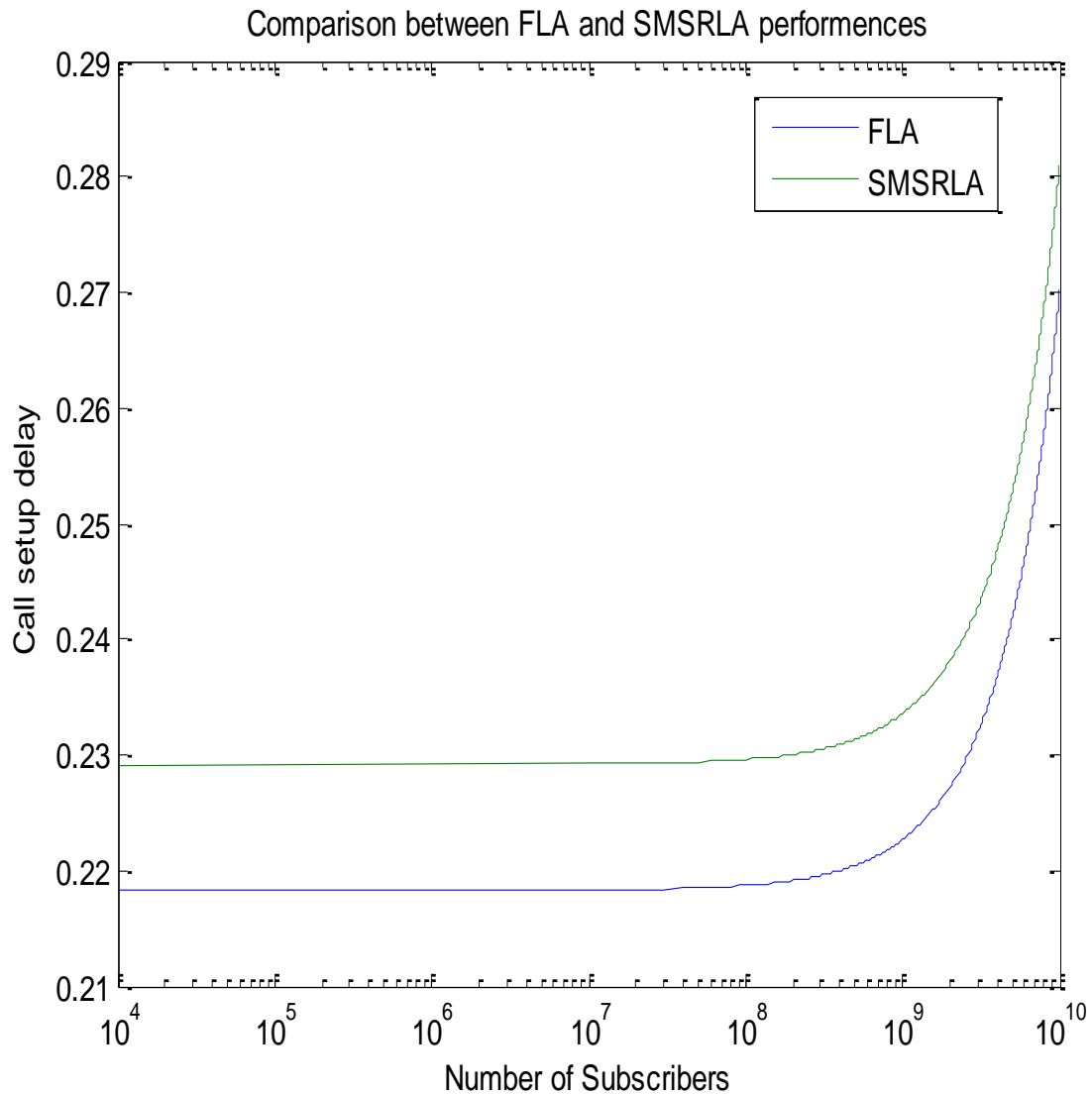


Figure 5.15 Comparison between FLA and SMSRLA performance.

6. CONCLUSION AND FUTURE WORK

This thesis discusses the Softswitch solutions for different networks. The implementation of softswitch solutions need to transform the telecommunication system from circuit-switching to packet-switching network. Utilizing Softswitch provides many of value-added-services (VAS) as well as the supporting of VOIP. As a result different type of services provided through gatekeepers and SIP proxy servers are controlled nowadays through Softswitch. There are many features of Softswitch such as IP based infrastructure and the separation of signaling and media information. Varieties of Softswitch networking architecture are presented in this thesis. The operation of softswitch could be handled in different planes. Those plans as well as their responsibilities and tasks are discussed in details.

The SS7 is the core part of telecommunication system for call control signaling. SS7 provides the fast and efficient service to exchange signaling information between switches. The SS7 protocol layers and their responsibilities are explained in one chapter.

Finally, the performance evaluation part of different types of softswitch architectures and some comparisons have been shown through simulations. Besides the theoretical analysis of the discussed softswitch architecture algorithms, we have presented several results based on simulations. Simulations show that when the subscribers ranging less than 7×10^5 the SLA has better performance. However, when the subscribers ranging more than 9×10^5 there are two networks architecture FLA and SMSRLA should provide better performance than SLA. Softswitch could work with different types of network infrastructure. Hence, it cannot stay only with existing structure but also possible to migrate to the NGN. Among this capability softswitch has become a powerful platform for network operator and they can develop other services as well.

The noticeable feature of this sector can be marked as price competition increasing highly among mobile operators; in this situation we can say that softswitch solution is a safe and cost effective solution for the current and near future networks. Softswitch provide important steps towards the future all- IP converged networks which support profitable traffic growth and enable new revenue-generating innovative services. Operators already have decreased their core network operating expenses by up to 50% deploying the solution and it can provide also innovative functionality beyond standardization. Cost-efficient way is most important in the competitive mobile market. Hence, mobile softswitch solution is a key step in the evolution of cost-efficient networks. It enables the migration of circuit switched voice traffic to an IP core network.

In future and as a continuation of this thesis, I will try to design better integration with existing network, considering the Load and Balancing for softswitch system. This provides the good performance for subscriber's storage and backup facilities. Also I will try to find out the delay of registration of session setup and the ratio of success calling.

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